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(71) Applicant
Lord Corporation,
1635 West 12th Street,
P.O. Box 2051, Erie,
Pennsylvania 16512,
United States of America

(72) Inventors
Glenn Edwin Warnaka,
Lyne Allen Poole,
Jiri Tichy

(74) Agents
Wynne-Jones, Laine & James,
22 Rodney Road,
Cheltenham,
Gloucestershire,
GL50 1JJ

# (54) Acoustic Attenuators with Active Sound Cancelling

(57) A microphone array 33 senses source sound in a duct 13 and converts it to an electrical signal which is sent to an electronic adaptive filter 23. The adaptive filter drives a speaker 17 for the production of cancelling sound 180° out of phase with the source sound, which propagates through a waveguide 19 to an acoustic mixer 15. A microphone 35 located in a position downstream from the waveguide 19 within the acoustic mixer 15 produces a signal which represents the error between the attenuation achieved in the acoustic mixer and the desired attenuation based on preset levels. This error signal is introduced into the adaptive filter 23 which then adjusts its signal driving the speaker 17 so that the cancelling sound propagating into the acoustic mixer 15 more nearly approximates the mirror image of the source sound. The adaptive filter has a modified LMS algorithm governing the operation of its transversal filter which accommodates the inherent delays associated with the propagation of acoustic waves in a duct. A plurality of wave guides may connect the speaker to the acoustic mixer, Figs. 6 and 7 (not shown). A single electronic circuit may provide the signal processing for a pair of ducts, Fig. 10 (not shown).

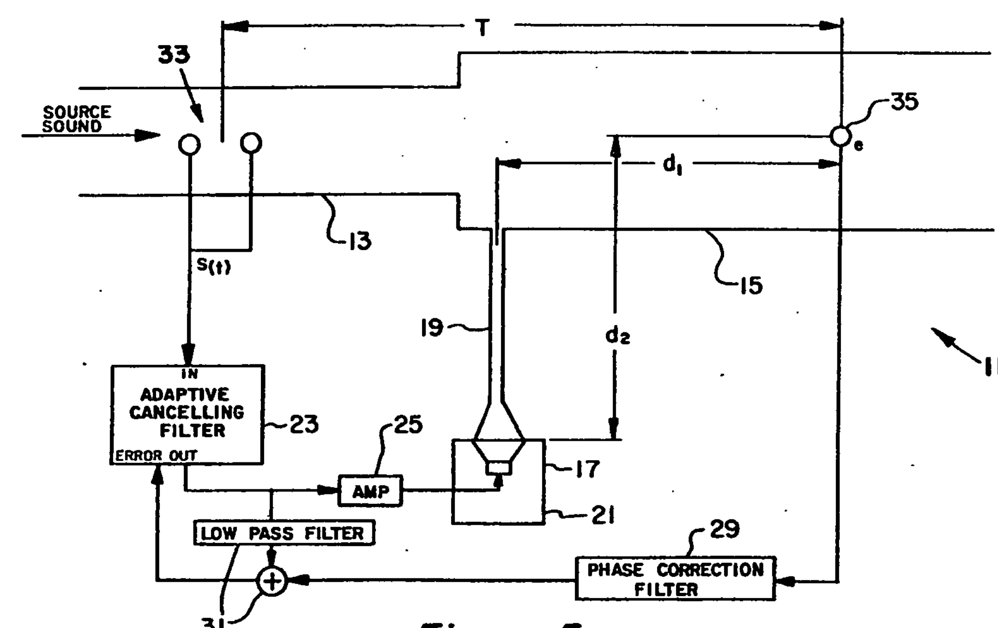



Figure 5

GB 2 088 951 A

PRIOR ART



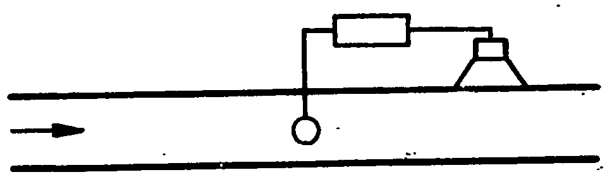


Figure 1

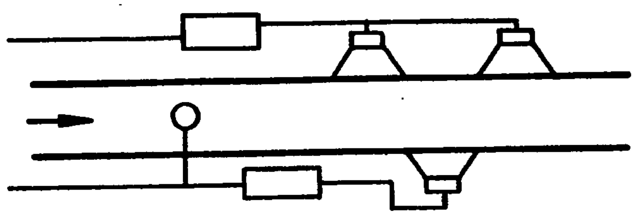


Figure 2

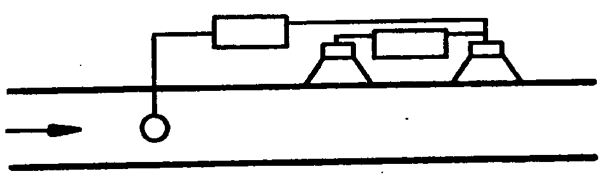


Figure 3

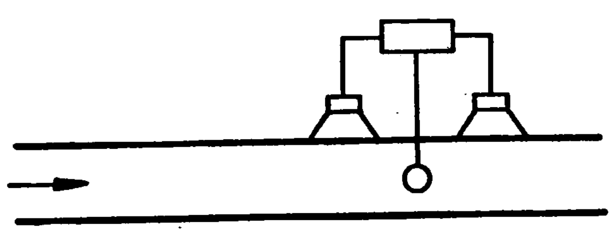


Figure 4

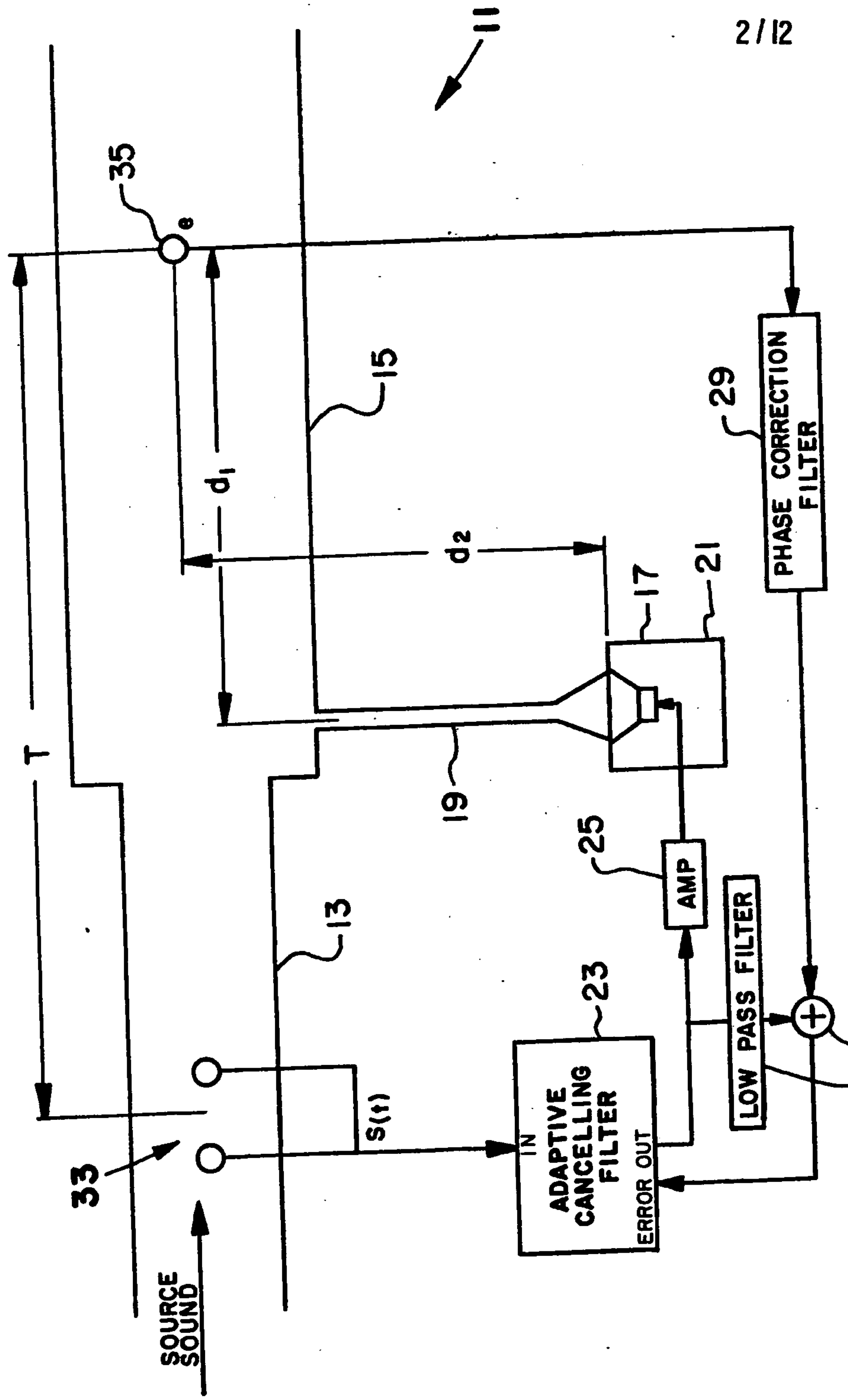

11
35
e
T
$d_1$
$d_2$
15
17
21
19
13
25
AMP
23
ADAPTIVE CANCELLING FILTER
IN
OUT
ERROR
S(t)
33
SOURCE SOUND
29
PHASE CORRECTION FILTER
LOW PASS FILTER
31

Figure 5

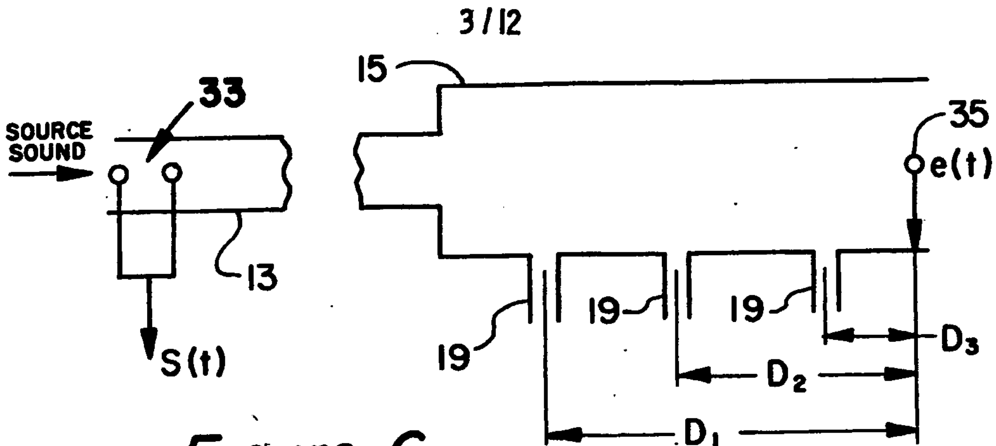



Figure 6

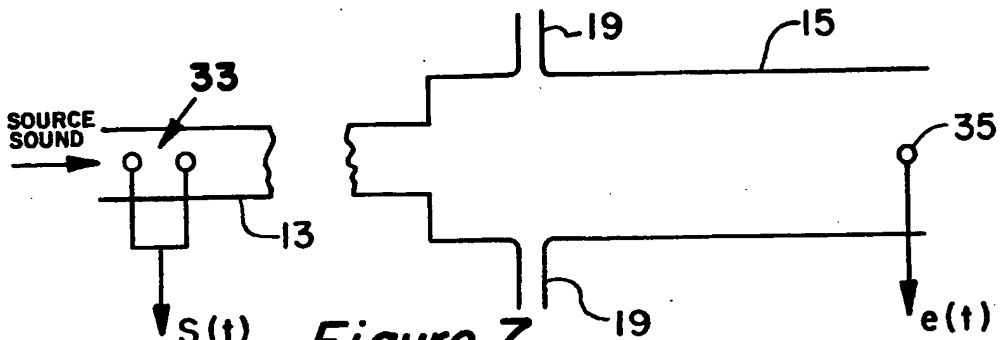



Figure 7

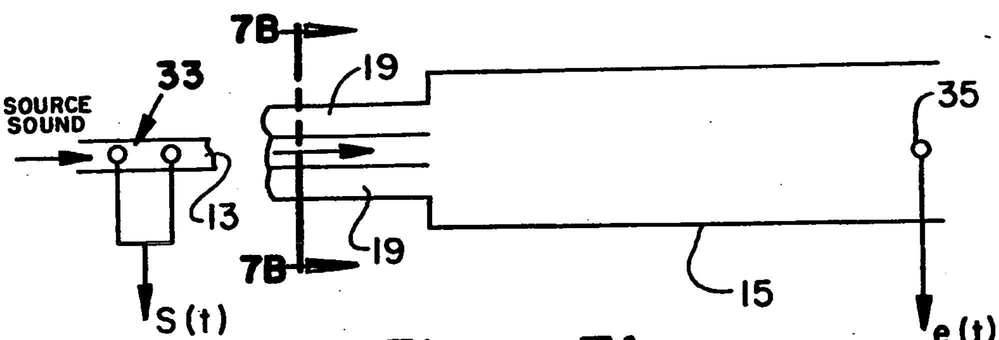



Figure 7A

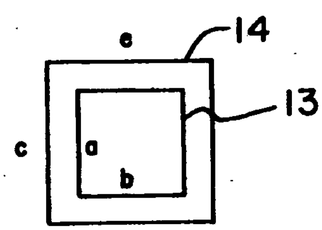



Figure 7B

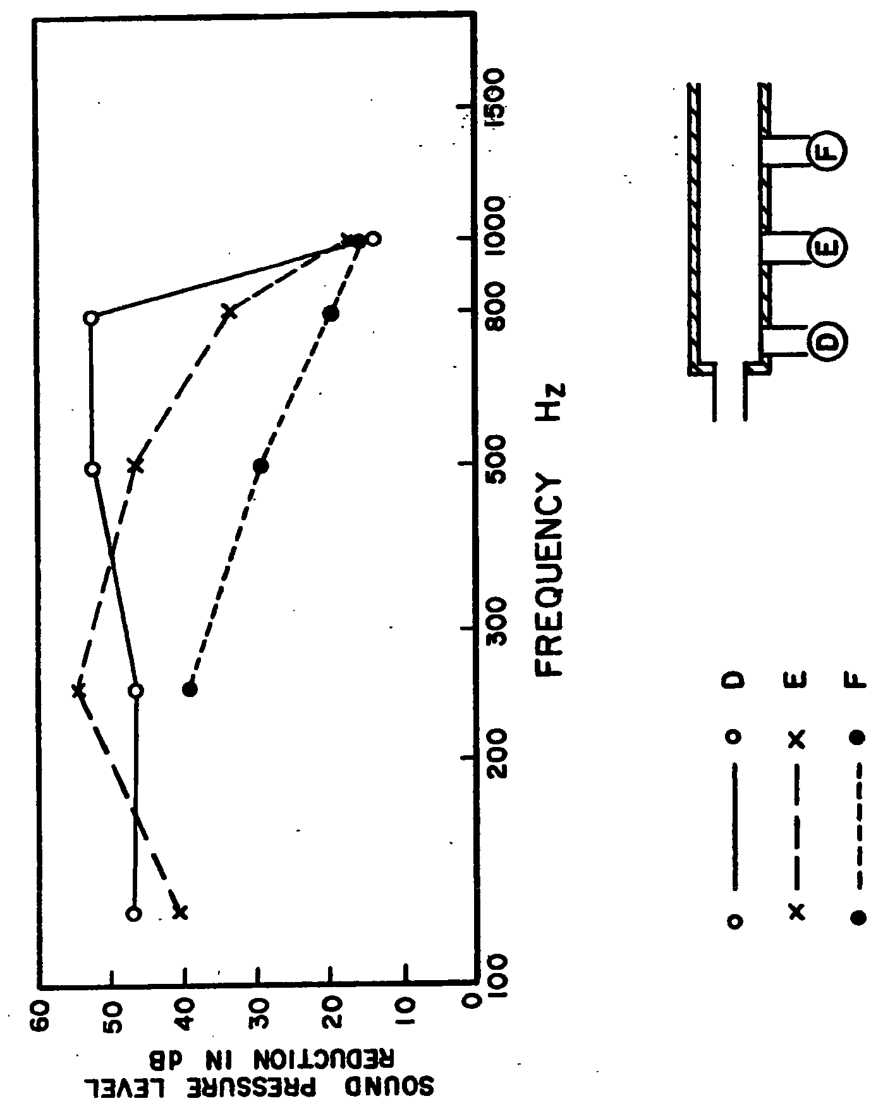

SOUND PRESSURE LEVEL
REDUCTION IN dB
0
10
20
30
40
50
60
100
200
300
500
800
1000
1500
FREQUENCY Hz
D
E
F
D
E
F

Figure 8

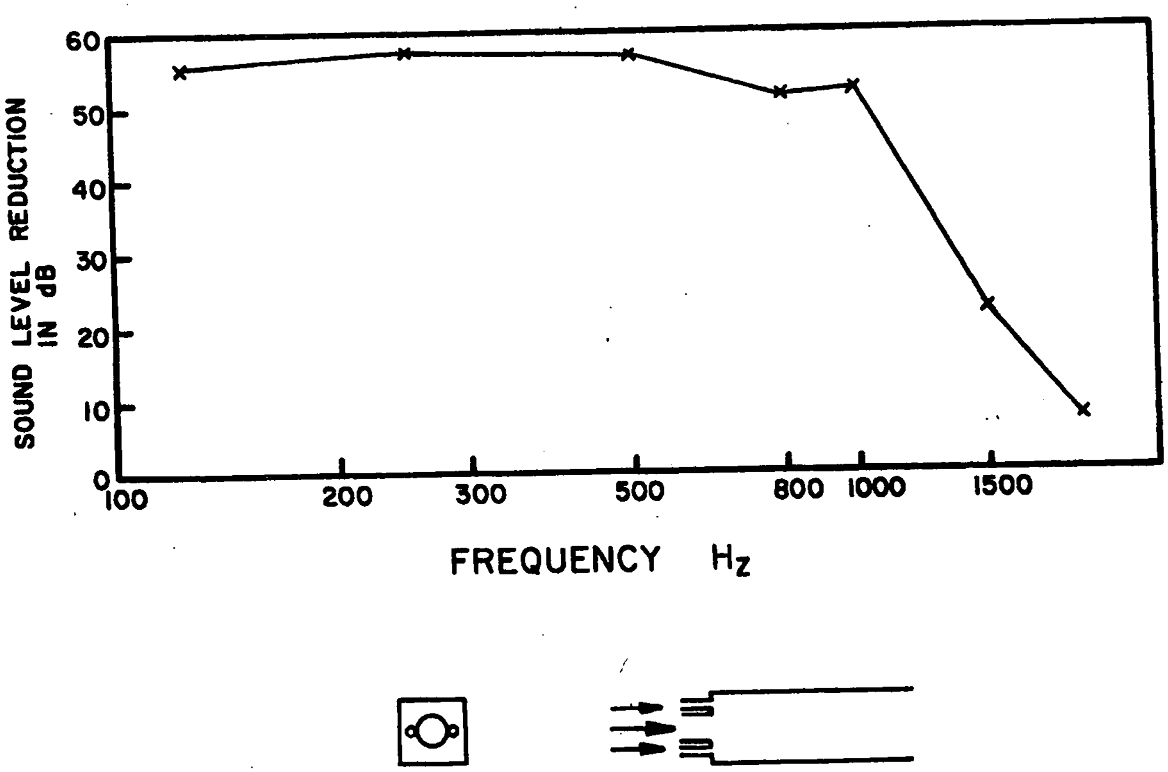

SOUND LEVEL REDUCTION IN dB
60
50
40
30
20
10
0
100
200
300
500
800
1000
1500
FREQUENCY Hz

*Figure 9*

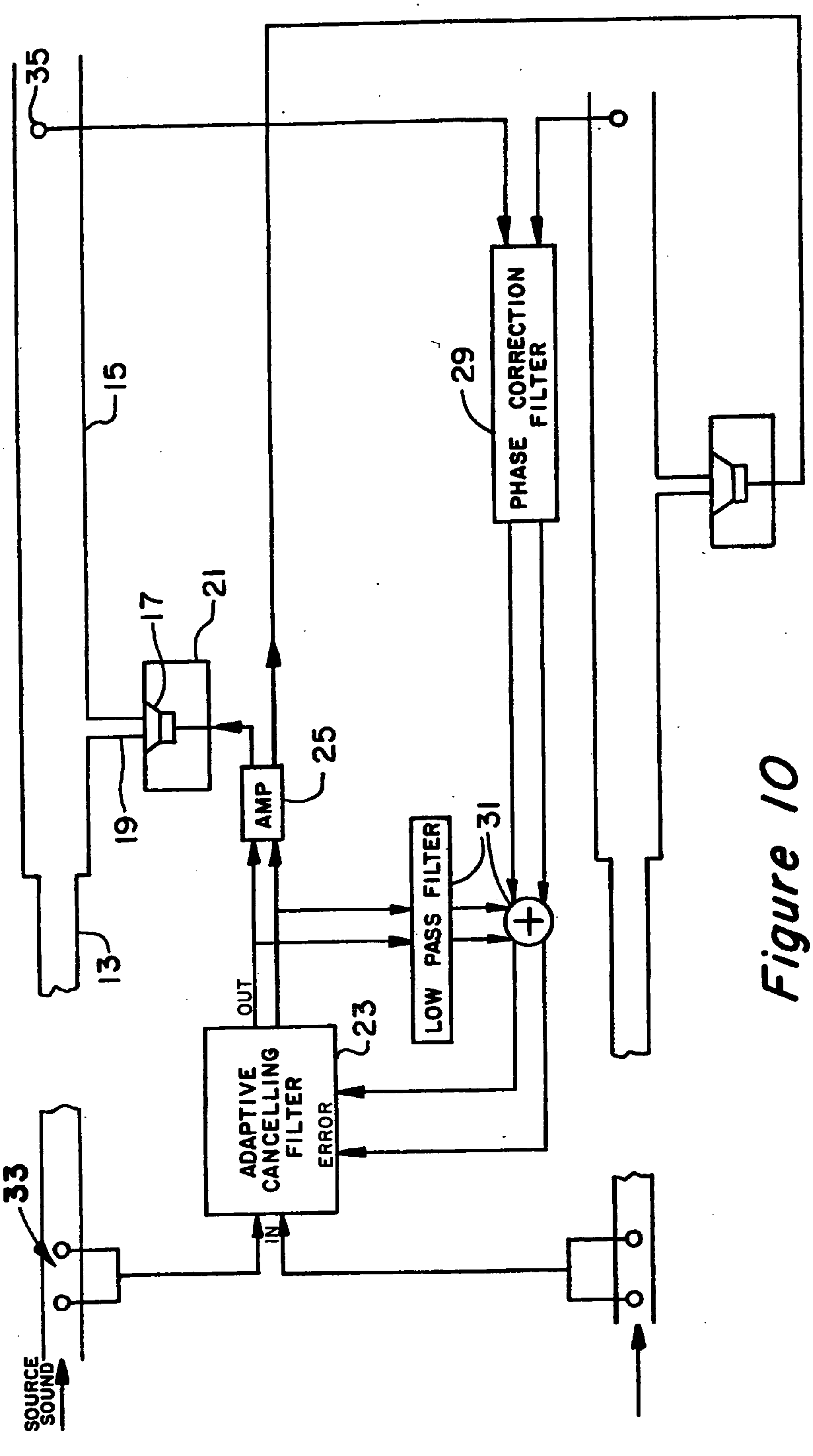

35
15
29
PHASE CORRECTION FILTER
17
21
25
AMP
19
31
LOW PASS FILTER
13
OUT
23
ADAPTIVE CANCELLING FILTER
ERROR
IN
33
SOURCE SOUND

Figure 10

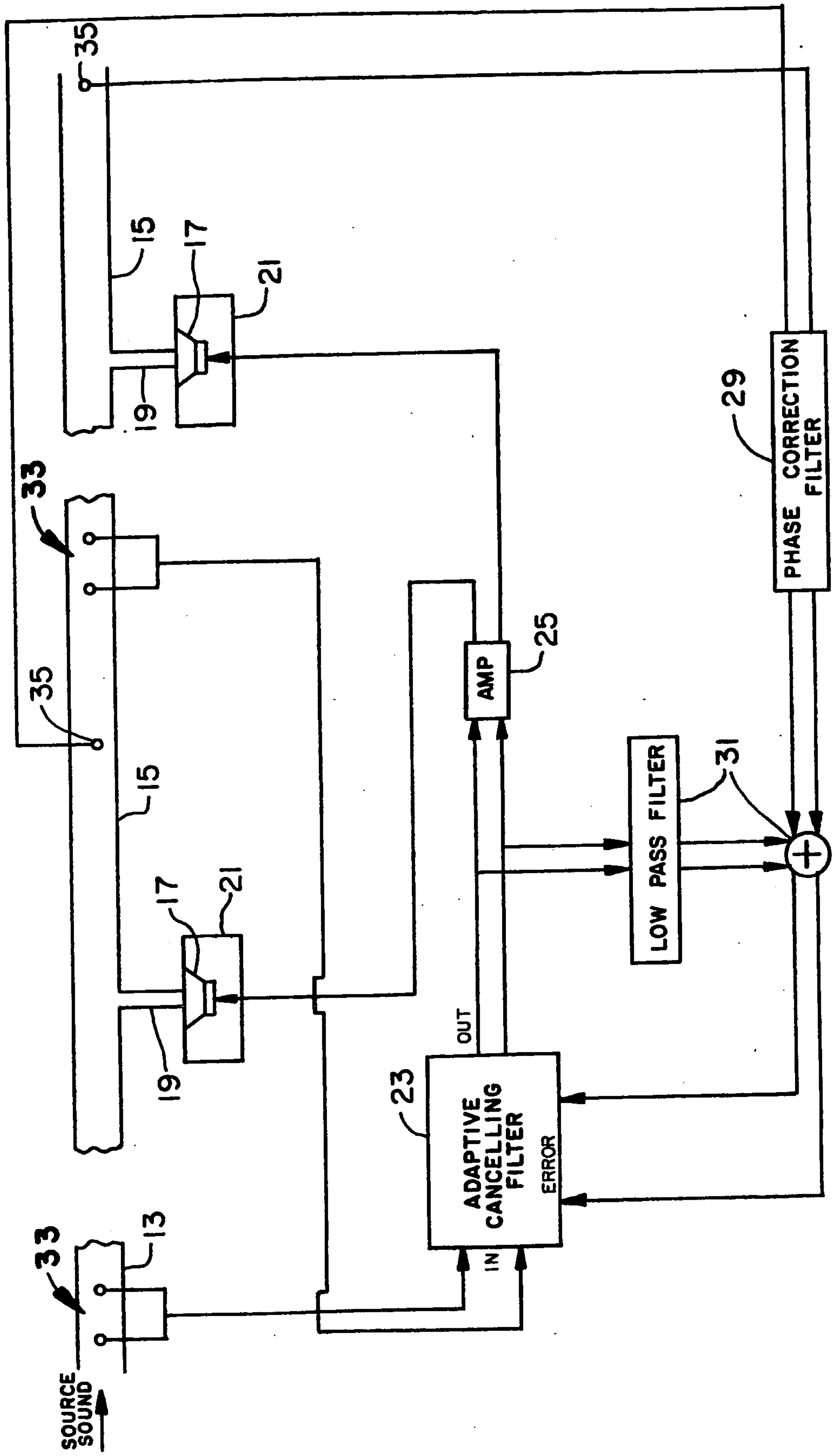

35
15
17
21
19
33
35
15
17
21
19
AMP
25
PHASE CORRECTION FILTER
29
LOW PASS FILTER
31
OUT
ADAPTIVE CANCELLING FILTER
ERROR
IN
23
33
13
SOURCE SOUND

Figure 11

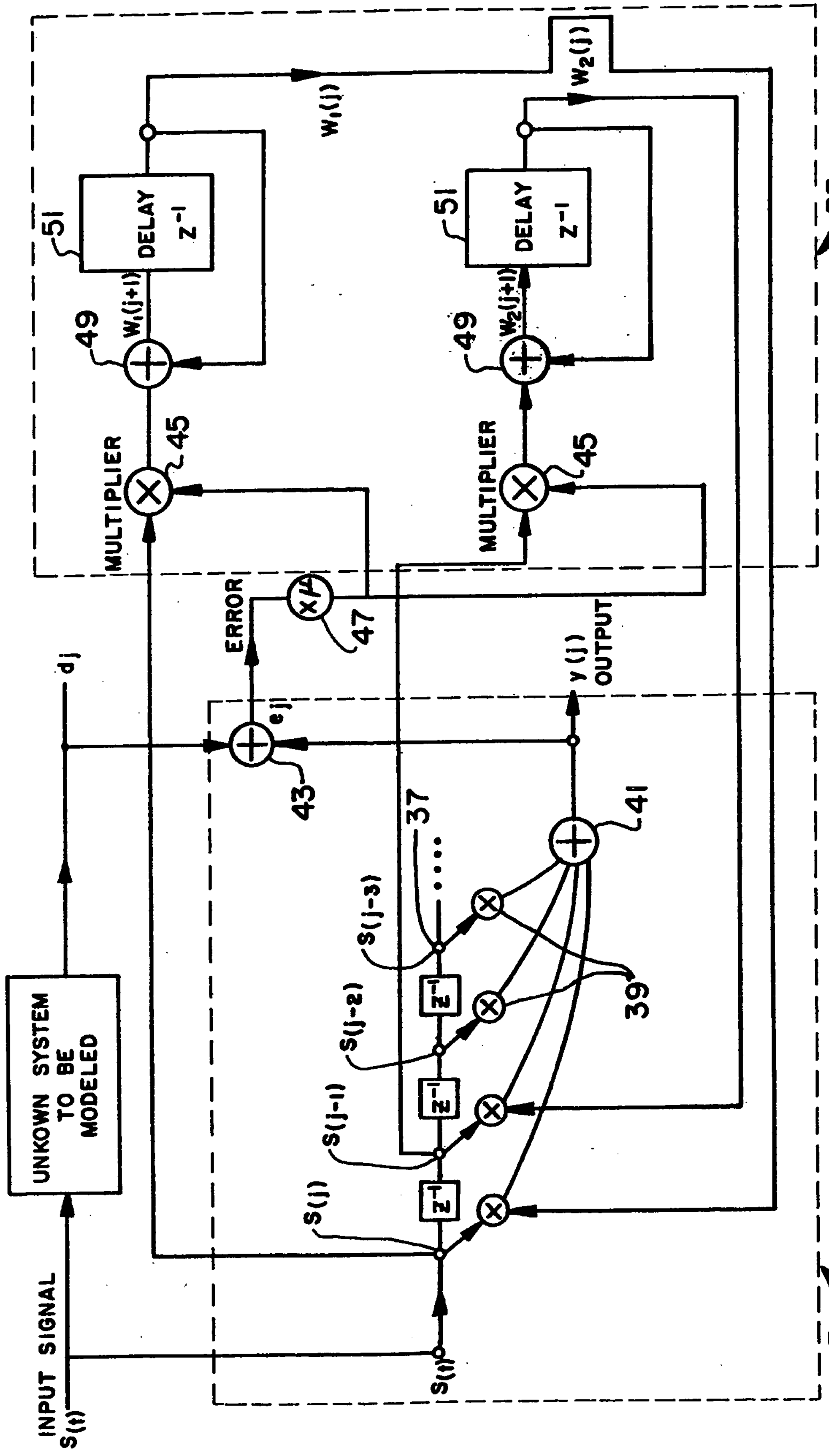

INPUT SIGNAL
S(t)
UNKOWN SYSTEM TO BE MODELED
dj
ERROR
ej
y(j)
OUTPUT
MULTIPLIER
MULTIPLIER
DELAY
$z^{-1}$
DELAY
$z^{-1}$
$W_1(j)$
$W_2(j)$
$W_1(j+1)$
$W_2(j+1)$
X μ
S(t)
S(j)
S(j-1)
S(j-2)
S(j-3)
$z^{-1}$
37
39
41
43
45
47
49
51
53
55

Figure 12

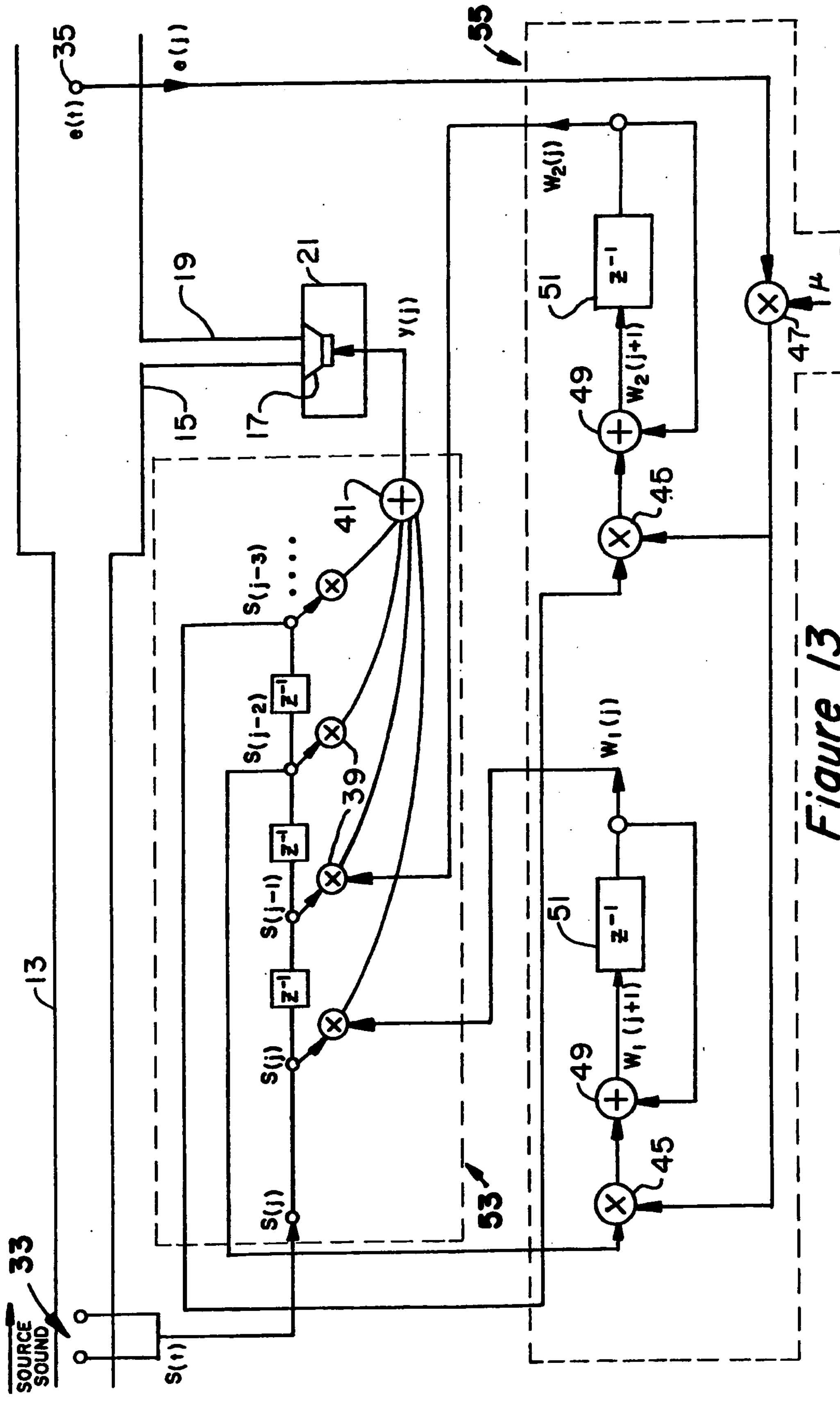

SOURCE SOUND
33
S(t)
13
15
17
19
21
35
e(t)
e(j)
Y(j)
41
39
S(j)
S(j-1)
S(j-2)
S(j-3)
$z^{-1}$
53
55
45
49
51
$W_1(j)$
$W_1(j+1)$
$W_2(j)$
$W_2(j+1)$
47
μ

Figure 13

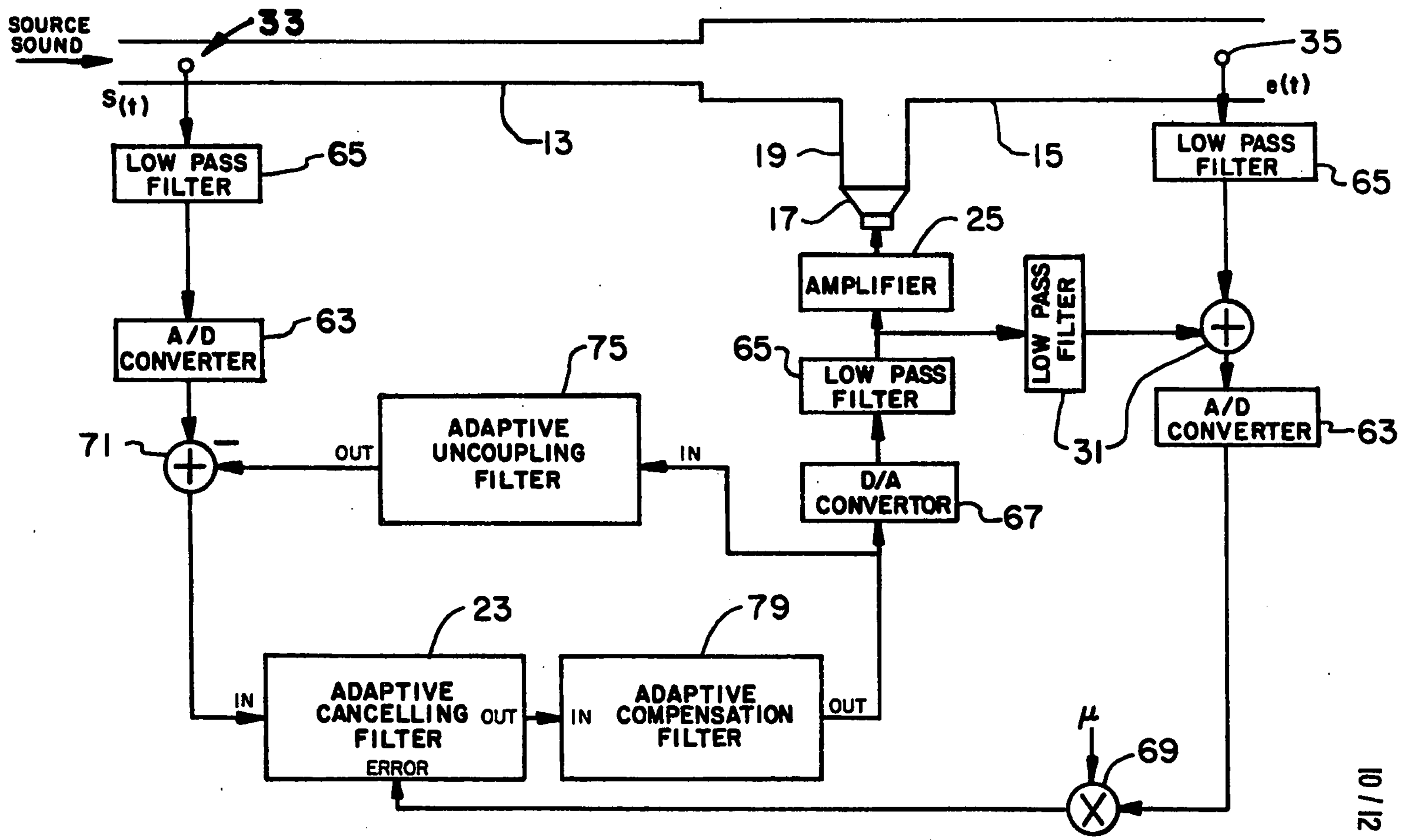

SOURCE SOUND
33
S(t)
13
19
17
15
35
e(t)
LOW PASS FILTER
65
A/D CONVERTER
63
71
ADAPTIVE UNCOUPLING FILTER
75
OUT
IN
AMPLIFIER
25
LOW PASS FILTER
65
D/A CONVERTOR
67
LOW PASS FILTER
31
LOW PASS FILTER
65
A/D CONVERTER
63
ADAPTIVE CANCELLING FILTER
23
IN
OUT
ERROR
ADAPTIVE COMPENSATION FILTER
79
IN
OUT
μ
69

Figure 14

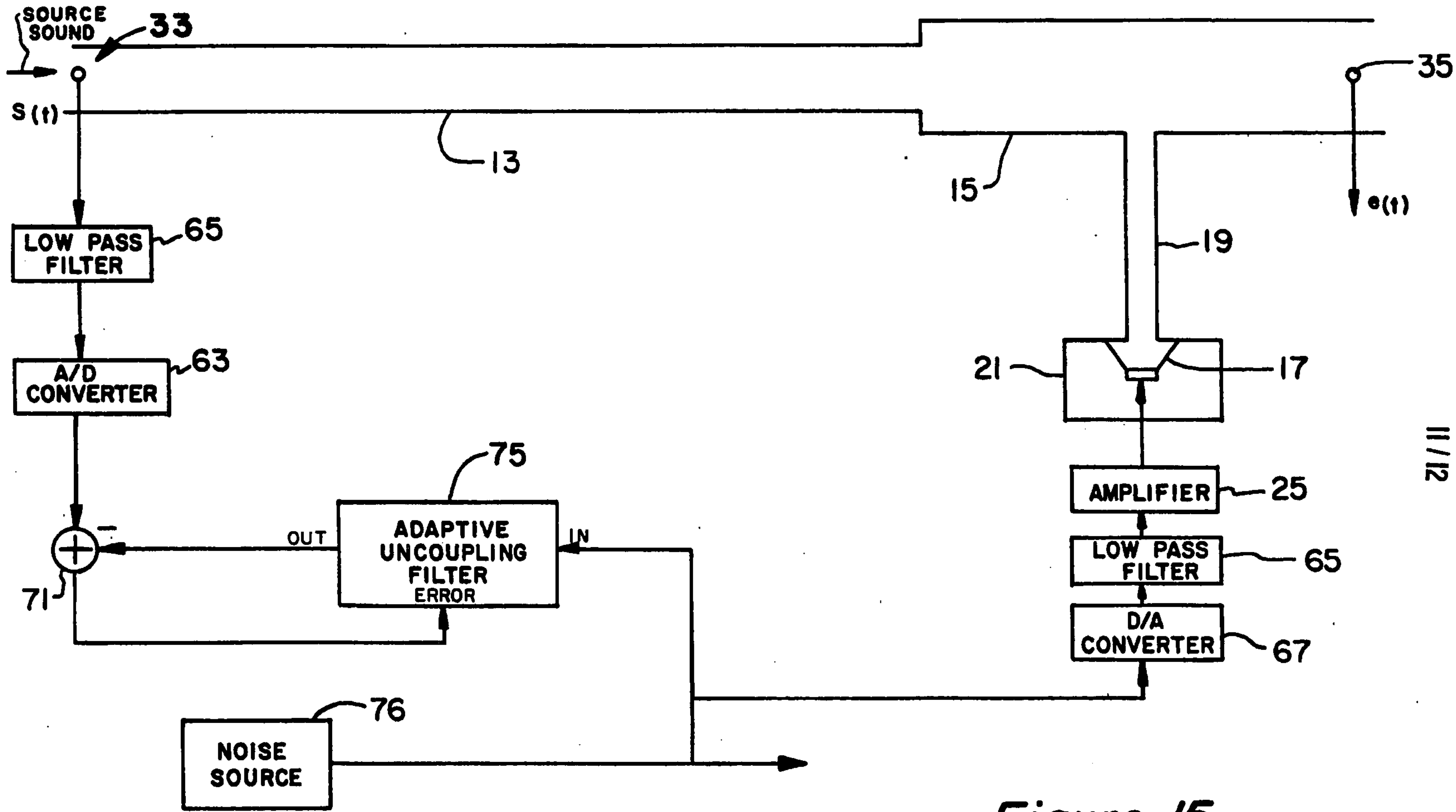
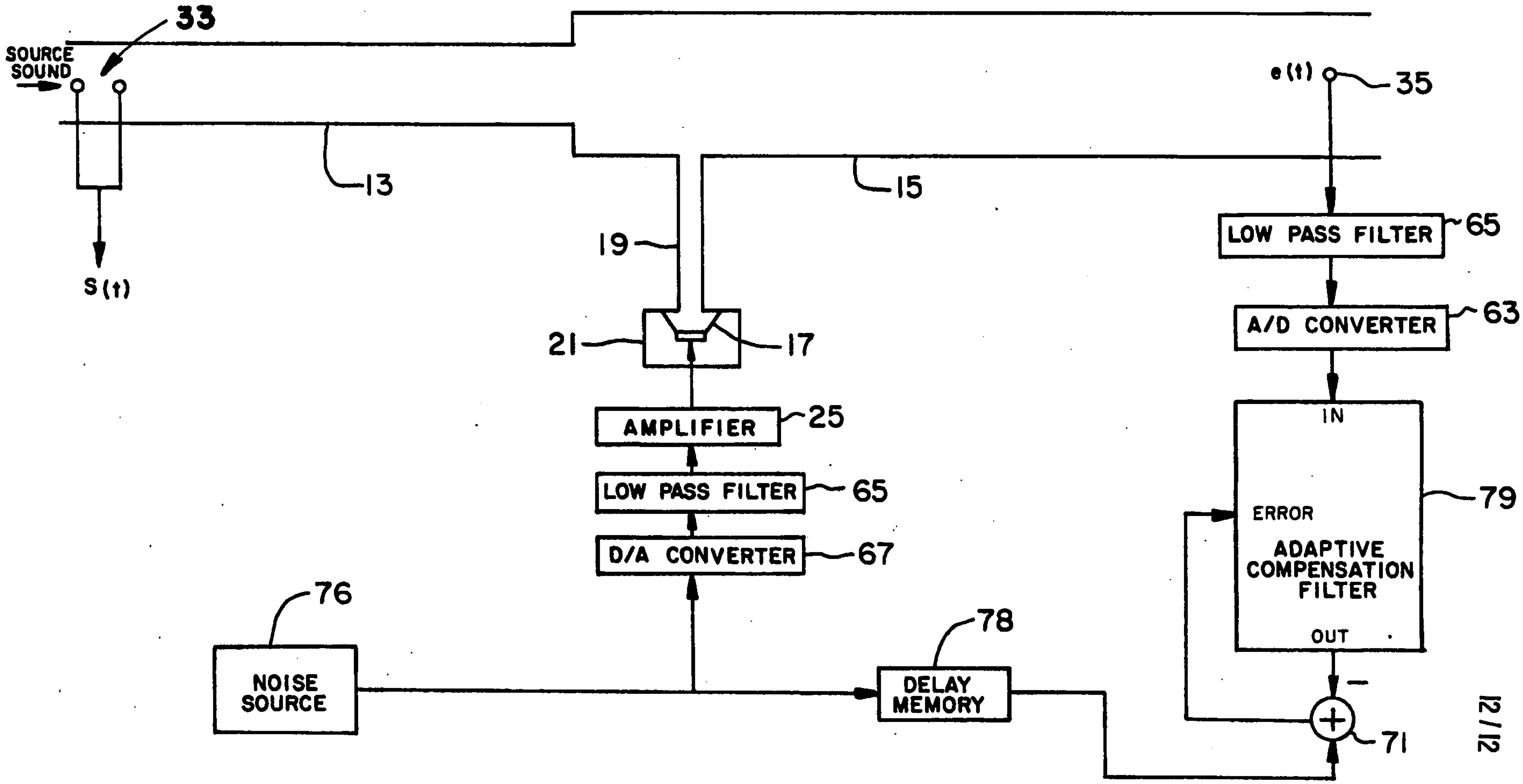

SOURCE SOUND
33
S(t)
13
15
19
35
e(t)
21
17
LOW PASS FILTER
65
A/D CONVERTER
63
ADAPTIVE UNCOUPLING FILTER
ERROR
OUT
IN
75
71
NOISE SOURCE
76
AMPLIFIER
25
LOW PASS FILTER
65
D/A CONVERTER
67

Figure 15



SOURCE SOUND
33
S(t)
13
15
19
21
17
AMPLIFIER
25
LOW PASS FILTER
65
D/A CONVERTER
67
NOISE SOURCE
76
DELAY MEMORY
78
e(t)
35
LOW PASS FILTER
65
A/D CONVERTER
63
IN
ERROR
ADAPTIVE COMPENSATION FILTER
OUT
79
−
71


Figure 16

SPECIFICATION

**Improvements Relating to Acoustic Attenuators**

This invention relates to the field of acoustic attenuation, and, more particularly, to a device capable of attenuation a narrow or broad band of lower frequency sound and single frequencies propagating from a given source through a confined area such as a duct by the introduction of cancelling sound which is 180° out of phase and equal in amplitude to the source sound.

Significant reductions in the sound pressure level of sound carried through confined enclosures such as ducts has been an unresolved problem for many years. In factories, for example, the noise produced by machinery and various manufacturing operations may be carried from the heating and ventilating ducts in such areas to the ducts which connect to offices and other parts of the plant in which a low level of noise is desired. This is particularly a problem with low frequency noise in the range of infrasound to 800 Hertz, since passive means to attenuate such frequencies are costly, relatively inefficient and physically large in size making them impractical for use in most low frequency applications.

Beginning in 1925 and continuing at an extremely rapid pace today, developments in electronics have made the concept of "active" attenuation of noise to be not only a possible but also an attractive alternative to passive attenuation of low frequencies. The principle of so-called active attenuation is based on the fact that the speed of sound in air is much less than the speed of electrical signals. In the time it takes for a sound wave to propagate from a location where it is detected to a remote location where it may be attenuated, there is sufficient time to sample the propagating wave, process such information within an electronic circuit and produce a signal to drive a speaker which introduces cancelling sound 180° out of phase and equal in amplitude to the propagating sound. Although the process of active attenuation of sound stated above appears to be conceptually simple, a review of the prior art in this area will illustrate the complexity of the problem and the difficulty of obtaining good attenuation over a relatively broad band of lower frequencies.

One of the first efforts in the area of active attenuators is a system as shown in Figure 1, comprising a monopole consisting of a microphone, amplifier and speaker. The microphone detects the source sound and converts it into an electrical signal which is fed to the amplifier. The loudspeaker, driven by the amplifier, is disposed downstream from the microphone at a location to give the necessary time delay to accomplish a 180° phase reversal from the source sound. The loudspeaker injects a mirror image of the source sound into the duct so that as the source sound passes the location of the loudspeaker, a volume of either high or low pressure air is introduced 180° out of phase with the corresponding high and low pressure volume of air of the source sound. When the loudspeaker is perfectly synchronized with the passage of the source noise, the pressure of the source noise and that of the loudspeaker averages to 0 (ambient static pressure) and the noise is "cancelled".

It is apparent from an examination of this system that attenuation will occur if the distance between the microphone (where the source sound is sampled) and the loudspeaker (where the cancelling sound is introduced) is such that the time delay of the electrical signal sent to the amplifier is equivalent to a 180° phase change or an odd multiple of 180°. However, this condition will occur only for a specific stationary acoustic signal which does not vary in time. As a practical matter therefore, this system is effective only for a single frequency since no means are provided to accommodate phase change. What this limitation shows is that there are two parameters which must be met for good attenuation, namely delay time, to allow the acoustic wave to move from the point of detection to the point of attenuation, and phase to assure that attenuation occurs at the point of introduction of the cancelling acoustic wave.

In addition to the limitation of the system shown in Figure 1 associated with phase detection and accommodation, a problem exists with the generation of standing waves by the loudspeaker in the upstream direction toward the microphone. Because of the standing wave pattern, the pressure of the sound field at the microphone is artificially nonuniform which means that at a given frequency the microphone may be located at a node or antinode of a standing wave. Therefore the cancellation signal produced by the speaker may be made to be too great or too little. As a result, the sound field may be amplified by the standing waves in such a way that the resulting propagation downstream from the speaker could be even greater than the sound produced by the source. In addition, the standing wave field could intensify the sound pressure in the duct and more sound could pass through the walls of the duct creating a secondary problem.

In an effort to avoid the above-identified standing wave problem and expand the frequency range of attenuation, several active attenuation systems have been developed subsequent to the system of Figure 1. One prior art system, shown in Figure 2, utilizes the combination of a monopole/dipole arrangement with the dipole being located on one wall of the duct and the monopole being located on the opposite side of the duct as shown. The dipole and monopole of this system are phased so that they add in the downstream direction and subtract in the upstream direction allowing a unidirectional propagation to be formed when the sources are balanced. It has been found however that the results obtainable with this system are frequency dependent and related to the half wave length of the dipole

separation. In addition, the complexity of this system does not lend itself for use in many practical applications.

A simplification which obtains improved performance, is shown by the dipole systems of Figures 3 and 4. The system shown in Figure 3 omits the monopole found in Figure 2 and the phase characteristics of the dipole are altered so that the propagation of both sources is added in the downstream direction and cancelled in a direction upstream toward the microphone. The active attenuator shown in Figure 4, locates the microphone between two loudspeakers to produce a minimum level at the microphone position when the proper phasing between the speakers is introduced. While this system is reflective and produces a standing wave pattern upstream of the dipole, the detection system (microphone) is not affected because it is located between the speakers, unlike the system of Figure 3.

In reviewing the performance of the dipole and the dipole/monopole systems, it has been found that each of these multisource systems seen to have geometry related limitations. The physical spacing of the loudspeakers and microphones produces a "tuning effect" which sets the frequency of best performance and the bandwidth. Although high levels of attenuation are possible, particularly with the system of Figure 3, such a performance is obtainable only through a relatively small band width of the order of about 2-1/2 octaves maximum. Accordingly, the most recent approaches to active noise attenuation in ducts have concentrated on improvement of the monopole system discussed above with reference to Figure 1. Thus in one example of a monopole system in which relatively sophisticated electronic circuitry is incorporated into the basic microphone/loudspeaker construction found in the system of Figure 1, a secondary cancelling wave is produced by a first electrical signal which represents the primary sound wave sensed by a microphone. The first electrical signal is convolved with the second signal derived from the system impulse response as a program of operational steps. This process represents the standard operation of an adaptive filter. To avoid the problem of standing waves produced by the loudspeaker and propagated upstream to the microphone, the use of a unidirectional microphone or lining the duct between the speaker and microphone with acoustically absorbant material has been suggested. In addition, the possibility of incorporating circuitry to provide a second convolution capable of compensating for the feedback signal produced by the upstream standing waves is suggested. However, as discussed in more detail below, the so-called adaptation process of such modified systems occurs over a period of from 5 to 30 minutes and is thus dependent on the source signal remaining essentially constant over that period. Moreover, the control system functions so slowly that in most practical cases it must be manually shut off while the first convolution process is still operative to avoid "system hunting" or oscillation between better and poorer results.

It should also be noted that each of the systems identified above share a common problem which in many practical applications will reduce if not eliminate their potential use. In each case, the speaker portion of the system is placed directly in the environment of the duct to introduce the cancelling sound. It is anticipated that in many applications existing speakers will not be able to survive the temperatures, particulates or other foreign materials found in the environment of the duct or in unprotected areas outside of the duct.

According to the present invention, in one aspect, there is provided an apparatus for the attenuation of sound from a source propagating in a duct comprising:

sensing means disposed in said duct for detecting said source sound;

cancelling means for generating cancelling sound of corresponding amplitude and 180° out of phase with said source sound, said cancelling means being disposed at a remote location from said duct;

waveguide means connecting said cancelling means with said duct, said waveguide means being connected to said duct at a location spaced from said sensing means in the direction of propagation of said source sound, said waveguide means providing a path for the propagation of said cancelling sound from said cancelling means to said duct; and

electronic controller means connected with said sensing means and being operable to activate and control said cancelling means for the production of said cancelling sound to attenuate said source sound within said duct and everywhere in the far field.

The preferred arrangement of the apparatus of the subject invention is a modified monopole-type active acoustic attenuator which may be thought of as consisting of three separate components namely a physical system, electronic circuitry and a coupling component between the physical system and circuitry. The physical system consists of the duct through which sound waves propagate from a given source, an acoustic mixer connected in line with the duct, a speaker disposed in a protective enclosure at a spaced distance from the duct and a waveguide connecting the speaker to the acoustic mixer. The electronic circuitry component of the active attenuator in this preferred arrangement consists of three distinct adaptive filters utilizing a modification of the Widrow-Roff LMS algorithm in a true adaptive acoustic cancelling configuration. A microphone array, disposed in the duct at a location upstream from the acoustic mixer, and a microphone positioned within the acoustic mixer downstream from the waveguide form the coupling means between the physical system and electronic circuitry.

As discussed in detail below, the operation of the preferred arrangement may be described in general terms as follows. The microphone array senses the source sound in the duct and converts it to

an electrical signal which is sent to a modified adaptive filter in the electronic circuitry component herein. The adaptive filter generates a signal to drive the speaker for the production of "cancelling" sound 180° out of phase with the source sound, which propagates through the waveguide to the acoustic mixer where it is combined with the source sound. The microphone located in a position downstream from the waveguide within the acoustic mixer, detects the sound resulting from the combination of the source and cancelling sound and produces a signal which represents the "error" or difference between the attenuation achieved in the acoustic mixer and the desired attenuation based on present levels. This error signal is introduced into the adaptive filter which then adjusts its signal driving the speaker so that the "cancelling" sound propagating into the acoustic mixer more nearly approximates the mirror image of the source sound. High levels of attenuation are thus achieved within the acoustic mixer and everywhere in the far field.

In its basic form the subject invention provides an active acoustic attenuator consisting of a physical component, an electrical circuitry component and coupling means between the two. Ideally the physical system component is characterised by a speaker disposed in a protective enclosure at a location remote from the duct, which is connected thereto by a waveguide, and includes an acoustic mixer into which a waveguide is connected at a discrete distance from the termination thereof.

A specific geometrical relationship between the relative location of the microphone array, speaker, waveguide and error microphone will achieve desired attenuation.

It is preferred that the active acoustic attenuator should have an electronic circuitry component characterised by adaptive filters utilising a modified Widrow-Roff LMS algorithm, and which will operate in a deterministic fashion, in contrast to prior art trial and error circuitry functions. Ideally the electronic circuitry component will consist of three distinct adaptive filters, each providing a separate function.

Several improvements and departures from prior art active attenuator systems have been made in the subject invention both in the physical system and electronic circuitry components described above. These will become apparent in the following discussion of the preferred embodiments. Such improvements have resulted in a performance capability of the active attenuator herein which far exceeds that of known systems both in terms of the attenuation achieved and the speed and flexibility of operation.

The invention may be performed in various ways and preferred embodiments thereof will now be described with reference to the accompanying drawings, in which:—

Figure 1 is the prior art monopole acoustic attenuator system;

Figure 2 is the prior art tripole or monopole/dipole acoustic attenuator system;

Figure 3 is the prior art dipole acoustic attenuator system;

Figure 4 is the prior art dipole acoustic attenuator system;

Figure 5 is a view of a simplified version of the active acoustic attenuator of the subject invention;

Figure 6 is a partial view of the subject invention showing various points of introduction of the waveguide into the acoustic mixer;

Figure 7 is an alternate embodiment of the subject invention in which two waveguides are symmetrically introduced into the acoustic mixer;

Figure 7*a* is a variation of the symmetrical introduction of the waveguide(s) into the acoustic mixer;

Figure 7*b* is a partial cross-sectional view taken generally along line 7*b*—7*b* of Figure 7*a*;

Figure 8 is a graph of the attenuation achieved herein with the configuration of the subject invention shown in Figure 6;

Figure 9 is a graph of the attenuation achieved herein with the configuration of the subject invention shown in Figure 7*b*;

Figure 10 shows a multistaging function of the subject invention in which a single electrical circuitry component is used with physical system and coupling components in two separate ducts;

Figure 11 shows a multistaging function of the subject invention in which two sets of microphone arrays, speakers and error microphones are disposed in series within a single duct controlled by a single electrical circuitry component;

Figure 12 is a partial block diagram of the digital realisation of an adaptive filter with the standard Widrow-Roff LMS algorithm;

Figure 13 is a partial block diagram of the digital realisation of an adaptive filter herein with a modified Widrow-Roff LMS algorithm;

Figure 14 is a block diagram of the electrical circuitry component of the subject invention;

Figure 15 is a block diagram of the adaptive uncoupling filter portion of the electrical circuitry component herein; and

Figure 16 is a block diagram of the adaptive compensation filter portion of the electrical circuitry component herein.

Initially, it should be noted that the approach taken in the subject invention to achieve active attenuation of noise is believed to be a significant departure from the approach followed in the development of prior art systems. All previous known designs of systems for the active attenuation of sound within a duct have proceeded initially with an attempt to solve the problem in its entirety. This

approach invariably involves a symbolic mathematical treatment of the subject which tends to result in theoretical generalities at the expense of obtaining specific design details.

The subject invention was conceived and designed with the recognition that active systems for the attenuation of noise in ducts involve three discrete component parts, namely the physical system, the electronic system (controller) and the coupling system between the two. This approach recognises the parallel nature of the system and the fact that system performance cannot exceed the performance of any component part. By independently developing and testing each component part of the system herein, it has been found that results were obtained much more quickly than in past experience. This is primarily due to the fact that any problems encountered in system operation were much more easily identified as being caused by a specific component of the system, thus eliminating guesswork and unneeded modification. Although such a procedure seems to be conceptually apparent, it is believed that progress in the area of active sound attenuation has been needlessly delayed to at least some extent simply by the unduly complex approach to the problem utilised by prior art inventors.

**Physical System Component**

Referring now to the drawings and in particular to Figure 5, the active acoustic attenuator of the subject invention is labelled generally with the reference 11. As discussed above, the active attenuator 11 may be thought of as consisting of three distinct components and the discussion in this section will be directed primarily to the physical system component with general references to the other components where necessary. The physical system includes a duct 13 through which sound from a given source propagates, an acoustic mixer 15 connected to duct 13, a speaker 17 which is the source of cancelling sound and a wave guide 19 which connects the speaker 17 with the acoustic mixer 15.

Although the acoustic mixer 15 is shown as having a slightly larger diameter than duct 13, this geometric relationship is not required for proper performance of active attenuator 13 and is shown herein for purposes of discussion and illustration only.

One of the immediately apparent advantages of the active attenuator 11 over prior art systems is that the speaker 17 is disposed at a remote location from the duct 13, and is connected to acoustic mixer 15 by an elongated waveguide 19 which may be provided with valve means (not shown) to prevent dust particles, caustic material or other debris flowing through ducts 13 from damaging speaker 17. As shown in Figures 1 to 4, each of the prior art active attenuators dispose the speaker directly in the duct where the internal and external conditions which could be encountered in many applications would quickly damage or ruin them. The speaker 17 is not only protected from the internal environment of the duct 13 and acoustic mixer 15 by waveguide 19, but it is contained in an enclosure 21 which protects speaker 17 from the external environment of the duct 13. Enclosure 21 must have a high transmisstion loss and may also be lined with acoustically absorbant material to further prevent the output of speaker 17 from propagating in a direction other than through waveguide 19. It should be understood that waveguide 19 is completely separate from duct 13 and acoustic mixer 15: that is, waveguide 19 is not a branch duct and thus no flow from the main duct 13 is carried through the waveguide 19. Waveguide 19 is provided solely for the purposes of carrying the cancelling noise from speaker 17 to acoustic mixer 15 and for isolating and protecting speaker 17 from the environment of duct 13.

The electronic component of the active attenuator 11 is shown in its simplest form in Figure 5 for purposes of the present discussion. A detailed description of the electronics of the subject invention will be provided below including an explanation of the complete circuitry utilised herein. The simplified version of the electronic component of active attenuator 11 includes an adaptive filter 23, an amplifier 25, a phase correction filter 29 and a DC loop labelled generally with the reference 31. The coupling component of the subject invention, which couples the physical system with the electronic system, consists of a microphone array 33 disposed in duct 13 in advance or upstream of waveguide 19 for sensing the source sound, and a microphone 35 disposed in acoustic mixer 15 downstream from waveguide 19 for purposes to become apparent below.

Generally, in the cancelling mode of operation, active attenuator 11 operates as follows. Broad band noise propagates down duct 13 and is sensed by microphone array 33 which produces a signal sent to adaptive filter 23. Adaptive filter 23 provides an output to drive speaker 17 which introduces "cancelling" noise through waveguide 19 into acoustic mixer 15. Since a sound wave consists of a sequence of compressions and rarefactions at a given phase and frequency, the pressure of such waves can be reduced or "cancelled" by generating a secondary wave having compressions and rarefactions equal in amplitude and 180° out of phase with the primary sound waves. The microphone 35 located downstream from waveguide 19 in acoustic mixer 15 senses the degree of attenuation or cancellation of the source sound after the sound waves produced by speaker 17 have been combined with it. The signal from microphone 35 is sent to adaptive filter 23 as an error signal, which, in effect, is an indication of the attenuation achieved within acoustic mixer 15. The adaptive filter operates to adjust its output depending on the character of the error signal, so that the speaker 17 is driven to produce "cancelling" sound which is more nearly equal in amplitude and 180° out of phase with the source sound.

Referring now to Figure 6, a geometrical requirement for optimum operation of active attenuator

11 is illustrated. In Figure 6, the waveguide 19 is shown entering acoustic mixer 15 at location $D_1$, $D_2$ and $D_3$ respectively from the end of what will be considered the entire duct system. In accordance with the relationship shown below equation (1), it has been found that the waveguide 19 must be positioned at a specific distance from the end of acoustic mixer 15 to obtain proper attenuation of the source signal before it leaves the duct system. This relationship is

$$D>31 \qquad (1)$$

Where:

D=distance of waveguide 19 from the end of the duct system.

1=the largest dimension (height or width) of the acoustic mixer 15 or the diameter of a round duct.

The relationship shown in Equation (1) expresses a threshold condition for the physical placement of waveguide 19 from the end of the duct system wherein optimum attenuation results. Equation (1) may be expressed in terms of attenuation as follows:

$$A_{dB}=\beta \tanh(\alpha 1) \qquad (2)$$

Where:

$A_{dB}$=attenuation in decibels

$\beta$=a limiting maximum attenuation for a given geometrical relationship

1=characteristic duct dimension, e.g. the diameter of a circular duct length of a side of a square duct longest side of a rectangular duct

$\alpha$=Any number zero or greater

Equation (2) reveals that the attenuation obtained with the active attenuator 11 varies according to the hyperbolic tangent of $\alpha 1$ as the location of waveguide 19 is moved relative to the end of the duct system which has been shown in Figure 6 as the termination of acoustic mixer 15. As mentioned above, the system performance of active attenuator 11 can be no better than the limitation of any of its components. The geometrical limitation of the physical system component discussed above must be observed to optimise the overall attenuation which can be achieved.

It should be noted that while Figures 5 and 6 show the waveguide 19 as entering acoustic mixer 15 at a right angle, this is not a requirement of the subject invention. Extensive experimentation has been conducted in which waveguide 19 is disposed at various angles relative to acoustic mixer 15 including angles of 0°, 30°, 45° and 60°. In each instance, the level of attenuation observed was essentially equal to that for the case in which waveguide 19 is perpendicular to acoustic mixer 15. This feature of the subject invention is particularly advantageous from the standpoint of installation, since it is anticipated that many applications may have space limitations which would require that the waveguide 19 be introduced into a duct at an angle other than 90°.

One of the primary attractions of active noise attenuators is their usefulness in attenuating sound of relatively low frequencies. Referring to Figure 8, a graph of the attenuation possible in the physical system of the subject invention having the configuration shown in Figures 5 and 6 is illustrated. The results obtained show the decay in attenuation as waveguide 19 is moved toward the end of the duct system, in accordance with Equations (1) and (2). The attenuation levels were observed in a laboratory environment in a duct having dimensions of 5 inches by 7 inches, without using the electrical circuitry component of the subject invention. As such, the data points reflect the maximum attenuation levels which can be obtained with the phycial system described above. Actual attenuation levels using the electrical components herein have been found to be in the range of between 25 and 35 db for 5 inch by 7 inch ducts and for ducts having a wide range of dimensions. This reduction in the level of attenuation is due to normal component tolerances, approximations and errors in the electronic system and because the signal to noise ratio of state-of-the-art electrical components is of the order of 60 db, all of which act to limit the overall system performance.

As can be observed in Figure 8, the embodiment of the subject invention shown in Figures 5 and 6 is capable of providing relatively high attenuation to a frequency level where the first cross mode is encountered. For a 5 inch by 7 inch duct, the first cross mode occurs at approximately 980 Hz. Although higher frequencies could be attenuated using passive means, the size, expense and inefficiency of passive attenuators limits their use in many applications. It was unexpectedly discovered, however, that the range of attenuation possible with active noise attenuator 11 could be expanded to frequencies up to the second cross mode by introducing the cancelling sound through waveguides 19 disposed in the symmetric configurations shown in Figures 7 and 7a. Figure 9 shows a graph for a configuration of active attenuators 11 essentially the same as that shown in Figure 7a in which a pair of waveguides 19 are introduced parallel to and on either side of duct 13, which confirms the elimination of the first cross mode.

The following description will provide a short discussion of the theoretical reasons for the elimination of the first cross mode where cancelling sound is symmetrically introduced into a duct carrying the source sound. Referring to Figure 7*b*, which is an end view of the embodiment of the subject invention shown in Figure 7*a*, the dimensions of the primary duct 13 are given as *a* and *b*

through which the source sound is propagated, and the dimensions of the symmetrically disposed secondary duct 14 (corresponding to waveguide 19) are given as *c* and *e* through which the cancelling sound is propagated. Assuming that the source sound and cancelling sound propagate in the z direction, the sound pressure at any point (x, y, z≥0) in the duct may be expressed as follows:

$$P\ (x,y,z,t) = \sum_{m,n=o}^{\infty} (A_{mn})\ \left(\cos \frac{m\pi}{a} x \cos \frac{n\pi}{b} y\right) \cdot (e^{-jk_z z})\ (e^{j\omega t}) \qquad (3)$$

Where

$A_{mn}$: describes a coefficient dependent on the source

$$\cos\frac{m\pi}{a} x \cos\frac{n\pi}{b} y:$$

Describes the propagation of waves in the x and y direction for a duct with hard walls

$e^{-jk_z z}$: describes the propagation of waves in the z direction

$e^{j\omega t}$: describes a time term

From Equation (3) conventional methods of solution may be used to show that the sound pressure below the first cross mode may be expressed as:

$$Poo=\frac{\rho c}{ab}(U_1S_1+U_2S_2)e^{-jk_z z}e^{jt} \qquad (4)$$

Where:

$\rho$=density of the medium

c=speed of sound

$U_1$=velocity amplitude of the cancelling sound

$U_2$=velocity amplitude of the source sound

$S_1$=area over which the velocity amplitude is $U_1$

$S_2$=area over which the velocity amplitude is $U_2$

To achieve theoretically complete cancellation (Poo=0) Equation 4 becomes:

$$U_1S_1+U_2s_2=0 \qquad (5)$$

or, expressed differently:

$$U_1=-U_2\frac{S_2}{S_1} \qquad (6)$$

Equations 5 and 6 reveal that the volume flow of the cancelling source must have the same magnitude as that of the noise source but 180° out of phase, which is well known. However, the effect on the first cross mode of introducing the cancelling sound symmetrically to the source sound is surprising, and was first found emperically. Based on the theoretical considerations expressed above, it can be shown that the pressure at any point (x,y,z≥0) in the duct is given by:

$$P = \sum_{m,n=o}^{\infty} \frac{\omega\rho}{k_z} (B_{mn})\ \left(\cos \frac{m\pi}{a} x \cos \frac{n\pi}{b} y\right) \cdot (e^{-jk_z z})\ (e^{j\omega t}) \qquad (7)$$

For the conditions of cancellation of plane waves given in Equations (5) and (6) above, the coefficient $B_{mn}$ can similarly be shown by conventional techniques to be:

$$B_{mo}=\frac{2}{m\pi}\quad (U_1-U_2)[1+(-1)^m]\sin m\pi\frac{c}{a} \qquad (8)$$

Thus, for the first cross mode where m=1 and n=o, assuming *a* is greater than *b*, $B_{mo}$=0 since the

term $[1+(-1)^m]$ in Equation (8) equals zero for m=1. This shows that in the configuration of the subject invention shown in Figures 7 and 7a, the frequency range over which relatively high acoustic attenuation may be obtained is effectively doubled by elimination of the first cross mode.

Techniques similar to those used in time-sharing computers may be applied to the electronic circuitry component herein, with appropriate modification for microphone and loudspeaker transducer characteristics, to permit multi-stage or duplex operation of active attenuator 11 as shown in Figures 10 and 11. Referring to Figure 10, a separate microphone array 33, speaker 17 waveguide 19 and microphone 35 is provided for two separate ducts 13 each having an acoustic mixer 15. A single electrical circuitry component of the subject invention, as described above, is capable of controlling the separate physical system components to achieve attenuation in both of the ducts 13. Thus, in applications where two separate duct systems are run parallel to one another with each system carrying the same or a different source sound, the active attenuator 11 herein has the capability of achieving attenuation in both systems with a single electrical circuitry component.

A second application of this multi-stage or duplex capability of the subject invention is shown in Figure 11. In some instances it may be desirable to obtain a level of attenuation in excess of 25 to 35 db. The configuration of Figure 11 includes two sets of microphone arrays 33, speakers 17, waveguides 19 and microphones 35 with one set being disposed downstream from the other within the same duct 13. As the source sound propagates through duct 13 it will be attenuated by the first speaker 17 as discussed above, and then the attenuated source sound will be attenuated further by the second speaker 17 in the same manner. By placing two separate physical system components of active attenuator 11 in series within a single duct 13, the attenuation obtained will be additive and result in a significant reduction in the decibel level of the sound which finally leaves the duct system. Again, a single electrical circuitry component of the subject invention using time-sharing techniques is capable of controlling both sets of physical system components.

**Electronic Circuitry**

The second primary component of the active acoustic attenuator 11 of the subject invention is the electronic implementation or control of the physical system described above. A brief description of the prior art may be helpful in appreciating the advances made in the electronic control circuitry herein. The linear summation of two equal and opposite signals has long been recognised as one approach to producing an electronic or acoustic null. Several of the acoustic cancellers developed to date are based on the "equal and opposite" principle wherein a cancelling wave generated by a loudspeaker is introduced into a confined space such as a duct to reduce the pressure variations produced by sound waves propagating through the duct from a given source. In the simplest model, the signal generated by the source is considered to be a pure tone represented by a single rotating vector. The cancelling signal must track the source within some maximum permissible error to obtain the desired attenuation. A significant problem with this approach, which is recognised and solved by the present invention, is that the permissible amplitude and phase errors must be held within tolerances which are available only in the very best acoustic devices. For example, to obtain 20 decibels of cancellation the errors in the microphones, speaker and electronics of such systems must be less than 1 decibel in amplitude and 6° in phase. As discussed in more detail below, the subject invention overcomes this problem using a tolerance relaxation technique based on feedback principles. Rather than requiring accuracy in all components, a feedback system concentrates the performance requirements in a few easily controlled devices.

One of the most recent so-called active attenuation systems is found in U.S. Patent No. 4,122,303 to Chaplin et al, which ostensibly uses an "adaptive" process to generate a cancelling wave capable of creating a null when combined with the sound waves from a source. An examination of this system however reveals that the electronic circuitry does not rely on a true adaptive process, but involves a trial and error approach in which a series of successive guesses or estimations are made of the error signal, which is the difference between the desired and actual attenuation. Eventually, the guesses of the error signal become closer and closer approximations of the desired error according to preset values. Not only is the trial and error method unduly lengthy (in the range of between 5 and 30 minutes), but after the process is completed the system must be manually shut down to avoid "system hunting" or oscillation between better and poorer results. In addition, once the trial and error process has begun in the Chaplin et al system, any change in the source sound in the 5 to 30 minutes required to complete the process will also result in "system hunting".

One key aspect of the electronic circuitry herein is that it is a deterministic system utilising true adaptive filters in contrast to the trial and error approach taught in Chaplin et al. This means that the electronic circuitry of the subject invention automatically adjusts its own parameters and seeks to optimise its performance according to specific criteria. Additionally, by using the feedback principles mentioned above, the electronic circuitry herein is not nearly as dependent on the tolerance requirements of acoustic devices used in many of the known prior art systems. In fact, where the permissible amplitude and phase errors of acoustic devices used in prior art systems are on the order of 1 decibel in amplitude and 6° in phase, the electronic circuitry herein can tolerate amplitude and phase errors in the range of at least 10 decibels and 45° phase. By relaxing the tolerance requirements, the

installation and maintenance of this system may be performed by technicians of ordinary skill making the active acoustic attenuator 11 commercially viable in a variety of applications.

The electronic circuitry for the active attenuator 11 utilises a modified form of the Widrow-Roff LMS adaptive algorithm in a true adaptive acoustic cancelling configuration. The LMS algorithm was designed for use in signal enhancement systems where the signal to noise improvement, or the noise reduction, was achieved solely in the electronic circuitry. This algorithm was significantly modified in the subject circuitry to permit operation when the vibration or noise reduction and/or cancellation is to be achieved in a physical system having inherent delays such as in an acoustic field. The modifications of the Widrow-Roff LMS algorithm retain the signal processing advantages inherent in the original algorithm and permits these advantages to be applied to active acoustic cancelling problems. Other adaptive algorithms exist for the solution of the quadratic error function and many of these could be modified for satisfactory operation in an acoustic cancellor. The purpose of the adaptive algorithm, as discussed in detail below, is to find an optimum or near optimum solution to the cancelling filter problem. Such other algorithms, modified in accordance with the teachings herein, can accomplish this function and should be considered within the sccpe of the subject invention.

Before discussing the modification of the LMS algorithm in the adaptive filters of the subject invention, a review of an adaptive filter having a standard LMS algorithm governing its operation will be discussed. Referring to Figure 12, an adaptive filter using the standard Widrow-Roff LMS algorithm is shown. The basic element of an adaptive filter is known as a transversal filter 53 as indicated in dotted lines in Figure 12. The transversal filter 53 can be visualised as a series of delay elements with the filter output being derived from a weighted summation of the delayed outputs. In Figure 12, a set of n measurements S(t) is sampled to form n sample measurements S(j) where j is the sampled time index. Each of the points labelled 37 in Figure 12 can be considered as constituting sampled input values, with the $Z^{-1}$ factor representing a delay. Each sample value 37 is multiplied by a corresponding weighting coefficient W(j) in multiplier 39, and the weighted measurements are entered into a summer 41 to form an output $y_j$ which is the output of the transversal filter 53. This output $y_j$ is compared with a desired response $d_j$ in a summer 43 to form an error signal $e_j$.

The objective of the LMS algorithm which governs the operation of the transversal filter 53, is to deterministically obtain the weighting coefficient in such a way as to minimise the error signal $e_j$ and find the weighted sum of the input signals that best matches the desired response. Changes in the weight vector to accomplish this end are made along the direction of the estimated gradient vector based on the method of steepest descent on the quadratic error surface. A detailed treatment of this subject can be found in the article by Prof. Bernard Widrow entitled "Adaptive Filters" from *Aspects of Network and System Theory* edited by Rudolph E. Kalman and Nicholas DeClaris. A block diagram representation of a digital realisation of the LMS algorithm is found on the right-hand portion of Figure 12, which is contained in the dotted lines shown and labelled generally as 55. Although shown in digital form, it should be noted that an analogue realisation of the LMS algorithm and transversal filter may also be used.

For purposes of illustration, only two sample input values and their corresponding weighted functions are shown in the drawings. The input S(j) is sent to a multiplier 45. The error signal $e_j$, together with a scaling factor $\mu$, which controls the rate of convergence and stability of the algorithm, is entered into a multiplier 47. The scaled error signal is then multiplied in multiplier 45 with signal S(j), and that product is introduced to a summer 49 and unit daly 51 or C RAM (Coefficient Random Access Memory). The weight setting $W_1$ (j+1) is sent back to the adjustable weight 39 corresponding to input signal S(j+1) whose product then forms part of the output of the transversal filter 53. The same operation is conducted for input signal S(j—1) corresponding to weight $W_2$ (j). As mentioned above, the LMS algorithm has proved to be effective in many adaptive signal processing applications wherein the weight setting determined from the error signal can be fed directly back to the adjustable weight corresponding to the input signal for which it was determined with essentially no delay.

In the form shown in Figure 12, the LMS algorithm and transversal filter would not be suitable for use in an acoustic cancellation problem. The cancellation of an acoustic wave propagating down a duct requires an equal and opposite signal to interact with it to reduce the maximum pressure variations generated in the acoustic mixer 15. The interaction of the two waves requires a finite length of travel and a corresponding amount of time. Referring now to Figures 13 and 14, it can be observed that in the physical system of active attenuator 11 a finite length of time is required for an acoustic wave to propagate from the microphone array 33 where the input signal is detected. In addition, a delay exists in the time required for the cancelling sound produced by speaker 17 to propagate through waveguide 19 to acoustic mixer 15. These delays may be approximated as the distance in feet through duct 13, acoustic mixer 15 and waveguide 19 divided by 1100 feet per second, which is the speed of sound. The expected delay for most systems will be from a few milliseconds to a few tenths of a second. Using a sampling rate of more than 2000 per second, the delay expressed in sampling intervals will be from several to several hundred.

Expressed in terms of the sampling intervals, the delay may be given as follows:

Delay=K (1/sampling rate)

Where:

K=an integer constant

The Widrow-Roff LMS algorithm was modified to account for the inherent delay in the physical system of active attenuator 11 such that the weighting coefficients determined in the algorithm would be matched with the corresponding signal inputs in the transversal filter for which the weighting coefficients were determined.

For a new input signal sample S(j), the corresponding weighting coefficient may be expressed as follows:

$$W_1(j)=W_1(j-1)+\mu S[j-(K+1)]e(j-1) \quad (10)$$

Where:

$\mu$=scaling factor

The next value of the weights can be written in the following form:

$$W_i(j+1)=W_i(j)+\mu S(j-K)e(j) \quad (11)$$

Where:

i=tap identification

s(j—k)=input sample K intervals past

As each new sample value of the input signal is sent to the transversal filter 53 it operates to generate the product of the first weighting coefficient and the last input sample which is added to the product of the second weight and the next to last input sample, and so on until the last weight times the oldest sample is accumulated. The total accumulation of these terms is the transversal filter 53 output y(j). The weighting coefficients are updated by the corresponding input sample and error signal to account for the inherent delays in the physical system. Figure 13 shows a digital realisation of the modified LMS algorithm according to the subject invention (with K=2) in which the delay discussed above is accommodated so that corresponding weighting coefficients, input samples and error signals are combined in the adaptive filter to produce the output signal y(j).

Referring now to Figures 5 and 14, the electronic circuitry of the active attenuator 11 including adaptive filters utilising a modified Widrow-Roff LMS algorithm will be discussed. In its simplest form as shown in Figure 5, the basic operation of the electronic implementation of active attenuator 11 may be described by the following sequence:

1. Sample the source sound wave propagating down duct 13 (signal)
2. Delay, filter and scale the signal
3. Drive speaker 17 with the output of step 2, above, to inject the proper replica of the signal
4. Sense the acoustic output of acoustic mixer 15 (error)
5. Adjust (2) using the modified LMS algorithm
6. Return to (1)

The adaptive filter 23 shown in Figure 5 is a 3-port arrangement, consisting of 2 inputs and an output. It receives an input from the microphone array 33 which is an electrical signal corresponding to the source sound in duct 13, and produces an output signal to drive speaker 17 for the production of a mirror image replica of the source sound which is introduced in acoustic mixer 15 through waveguide 19. The output of acoustic mixer 15 sensed by microphone 35 is introduced to adaptive filter 23 as an error signal, or the summation of the source sound and the replica achieved in acoustic mixer 15.

The acoustic propagation delay inherent in the physical system of active attenuator 11 must appear as a delay over the full frequency range of interest. The phase tolerance of this delay in approximately ±45 degrees at each frequency. In the block diagram schematic of the electronic circuitry shown in Figure 5, a second-order phase-correction filter 29 is included to compensate for the acoustic resonances in the acoustic mixer 15. The characteristics of filter 29 must be determined manually, using appropriate instrumentation, based on the duct characteristics of a particular application. As discussed below, this function may be accomplished with an adaptive filter thus eliminating manual calibration of filter 29.

A DC loop is also provided in the block diagram of active attenuator 11 shown in Figure 5, which is labelled generally with the reference 31. Microphone 35 is not capable of detecting very low frequency components of the output from acoustic mixer 15, which, in addition to a DC component, are needed for stable operation of the LMS algorithm of adaptive filter 23. The DC loop 31 is included to provide the DC component, and the LMS algorithm has been modified to accommodate such input.

Referring now to Figures 14 to 16, an advanced form of the electronic component of active attenuator 11 is shown. Three discrete adaptive filters are included in the electronic component of active attenuator 11, each performing a separate function. Before discussing the adaptive filter processes, the function of the remaining circuit elements should be mentioned. A digital implementation of the adaptive algorithm is used in this embodiment of the electronic circuitry of active attenuator 11. The analogue to digital converters (A/D), labelled generally as 63 in Figures 14 to 16, provide the sample values of the selected inputs in digital format. The A/D converters 63 are 12 bit successive approximation converters.

Most of the signal processing within the electronic circuitry herein is digital, and the sampling rate of the A/D converters 63 implies an upper limit on the allowable bandwidth of the signal and error inputs. The sampling rate limit requires that the maximum input frequency must be less than half the sampling rate. The low pass filters 65 of the system inputs are required to limit the maximum input frequency to less than one-half of the A/D sampling rate. The output signal from the digital to analogue (D/A) converter 67, will also have a frequency aliasing problem. The output signal from the adaptive filters could excite a resonance in the system at any alias, or multiple, of the D/A converter 67 output. Therefore the low pass filter 65 on the output of D/A converter 67 is required to limit the maximum output frequency to the band of interest.

The multipliers 39, 45, 47 and 69, and accumulators or adders 41, 43, 49, and 71 shown in Figures 12 to 16 perform the computations necessary for the adaptive processing herein. It has been found that the TRW Model TDC1003J 12 bit Parallel Multiplier-Accumulator, or a suitable equivalent, provides the high speed calculation capabilities required.

The first of the three adaptive filters utilised in the electronic circuitry component of active attenuator 11 may be described as the adaptive cancelling filter 23 as shown in Figures 5 and 14. The adaptive cancelling filter 23 includes a modified LMS algorithm to govern the operation of its transversal filter as discussed above. To expand on the prior discussion, the basic filter functions of the adaptive cancelling filter 23, including phase response, amplitude and delay, are implemented by the transversal filter which is a non-recursive, finite impulse response filter. The filter implementation is digital in nature with both the input signal history and the tap weights being stored in a Random Access Memory (RAM). As mentioned above, the operation of the transversal filter can be described as generating the product of the first weight obtained from the LMS algorithm and the last input sample, plus the second weight from the algorithm times the next to last input sample, and so on until the last weight times the oldest sample is accumulated. In equation form, this may be given as follows:

$$\text{Output}=\sum_{i=1}^{i=I} S_i W_i \qquad (12)$$

Where:

$S_i$=sampled input signal corresponding to the ith tap
$W_i$=weight of the ith tap
i=tap identification
I=number of taps

The adaptive cancelling filter 23 may be considered to be a transversal filter when the weight adjustment are stopped. The operation of adaptive cancelling filter 23 will determine the required values for the weights of the desired filter (i.e., optimum filter conditions) by adaptive means and then realise the desired filter function by using the determined weights in the transversal filter. The purpose of the modified LMS algorithm is to govern the transversal filter operation by matching the filter function of adaptive cancelling filter 23 to the desired filter function.

The error signal sensed by microphone 35 and sent to adaptive cancelling filter 23, can be visualised as the summation of the source pressure wave sensed by microphone array 33 delayed by the acoustic travel from that point to microphone 35, plus the cancelling wave produced by speaker 17 delayed by the travel through waveguide 19 and acoustic mixer 15 to microphone 35. In physically positioning microphone array 33, speaker 17, waveguide 19 and microphone 35 relative to one another in the duct 13 and acoustic mixer 15, the total delay or length from microphone array 33 to microphone 35 must be greater than the total delay or length from speaker 17 to microphone 35 plus the delay associated with the operation of low pass filters 65. In equation form, this relationship may be expressed as follows (See Figure 5):

$$T>\frac{\lambda\text{min.}}{4}+d_1+d_2+D_f \qquad (13)$$

Where:

$\lambda$min=shortest wavelength of interest
T=distance between microphone array 33 and microphone 35
$d_1$=distance from waveguide 19 to microphone 35
$d_2$=distance from acoustic mixer 15 (i.e., in alignment with microphone 35) to speaker 17
$D_f$=delay associated with loss pass filters 65

The delay $D_f$ of the low pass filters 65 can be expressed in terms of cutoff frequency or $F_{max}$. A fourth order low pass filter 65 will have a delay of 45 degrees for each pole at the cutoff frequency, and a good filter design will approximate a constant delay filter. Therefore, the delay for a fourth order filter

can be approximated as $1/2F_{cutoff}$, and $D_f$ may be considered to be approximately equal to $1/F_{cutoff}$ for the two sets of fourth order filters in series.

The second adaptive filter found in the electric circuitry component of active attenuator 11 may be termed an uncoupling adaptive filter 75, as shown in Figures 14 and 15. One potential problem associated with most active attanuators is the production of standing waves or mechanical vibrations of the duct as a result of the cancelling sound introduced by the speaker propagating toward the microphone or microphone array which senses the source sound. This coupling will tend to corrupt the signal sensing microphone's estimate of the source sound propagating downstream and reduce the effectiveness of the cancelling system. Although unidirectional microphones have been suggested as a means to avoid this problem, they are not sufficient acting alone to overcome this inherent system limitation.

The uncoupling adaptive filter 75 herein solves this problem as follows. As shown in Figure 15, a broad band noise 76 drives the cancelling loudspeaker 17 and the uncoupling adaptive filter 75. Prior to start up of the system, the noise source 76 is automatically adjusted to drive the duct 13 at a sound level higher than the source level required to cancel the expected source noise. The adaptive process will reduce the output of the error summer and match the transfer function of the transversal filter in the adaptive uncoupling filter 75 to the acoustic coupling present in duct 13, at which time the noise source 76 will be terminated. At start up of the system, the adaptive uncoupling filter 75 will operate on the drive to the loudspeaker 17. The components of the loudspeaker drive that appear in the output from microphone array 33 will be removed by subtracting an equal and opposite image at summer 71. The subtraction process is accomplished electronically in the digital domain of the adaptive uncoupling filter 75 and in summer 71. Although this procedure will not entirely eliminate the effect of the output from speaker 17 on microphone array 33, sufficient reduction will be obtainable in most applications.

Referring now to Figures 14 and 16, the third adaptive filter of the electronic circuitry component in active attenuator 11 is shown, which may be called an adaptive compensation filter 79. As mentioned above in connection with the discussion of the basic electronic component shown in Figure 5. compensation means must be provided in series between the output of adaptive cancelling filter 23 and the input of the error signal thereto to assure stable operation of the modified LMS algorithm. While this compensation was shown as a second-order phase-correction filter 29 in Figure 5, it was noted that the manual calibration of filter 29 could be accomplished by an adaptive filter. The adaptive compensation filter 79 performs this function.

As shown in Figure 16, the adaptive compensation filter 79, loudspeaker 17, waveguide 19, acoustic mixer 15 and microphone 35 are in series with each other, and in parallel with the broad band delay circuit consisting of a delay or memory 78. Prior to start up of the system and following the completion of the uncoupling process described above, the noise source 76 will be activated to drive duct 13 and acoustic mixer 15. The summation of the parallel signal paths will be the error input into the adaptive compensation filter 79. The adaptive process will match the total transfer function of the series path to the true delay of the noise source 76 by delay 78. In this way the weights for the desired filter are generated, and the error signal received by adaptive cancelling filter 23 will be within the proper phase tolerance to assure stable operation.

Delay circuit 78 of Figure 16 will delay the signal from noise source 76 by an integer number of input sample intervals (K). The modification of the LMS algorithm in adaptive filter 23 in Figure 14, will provide for a delay of K sample intervals in the calculation of the next value for $W_i$. The value of K will be greater than the acoustic delay from loudspeaker 17 to the signal input of the adaptive compensation filter 79 of Figure 16. The value of K can be set large so that one value can be used in most applications. The matching of the delay from the output to the error input of the adaptive cancelling filter 23 in Figure 14, to the shift in the sampled signal used in the calculation of the next weight value, will assure the stability of the modified LMS algorithm.

While the invention has been described with reference to preferred embodiments, it will be understood by those skilled in the art that various changes may be made and equivalents may be substituted for elements thereof without departing from the scope of the invention as defined in the appended claims. In addition, many modifications may be made to adapt to a particular situation or material. Therefore the invention is not limited to the particular embodiments disclosed as the best known modes contemplated for carrying out this invention.

## Claims

1. An apparatus for the attenuation of sound from a source propagating in a duct comprising:

sensing means disposed in said duct for detecting said source sound;

cancelling means for generating cancelling sound of corresponding amplitude and 180° out of phase with said source sound, said cancelling means being disposed at a remote location from said duct;

waveguide means connecting said cancelling means with said duct, said waveguide means being connected to said duct at a location spaced from said sensing means in the direction of propagation of said source sound, said waveguide means providing a path for the propagation of said cancelling sound from said cancelling means to said duct; and

electronic controller means connected with said sensing means and being operable to activate and control said cancelling means for the production of said cancelling sound to attenuate said source sound within said duct and everywhere in the far field.

2. The apparatus of claim 1, wherein said sensing means is a microphone array.

3. The apparatus of claim 1 or claim 2, wherein said cancelling means is a speaker.

4. The apparatus of any one of claims 1 to 3, wherein said electronic controller means includes an adaptive cancelling filter, an amplifier, a phase correction filter and a DC loop means, said adaptive cancelling filter including a transversal filter and a modified LMS algorithm governing the operation thereof, said adaptive cancelling filter accommodating the inherent acoustic time delays of said apparatus resulting from the time required for propagation of said source sound through said duct and of said cancelling sound through said waveguide.

5. The apparatus of any one of claims 1 to 4, wherein said cancelling means is disposed in a protective enclosure lined with acoustically absorbant material.

6. The apparatus of any one of claims 1 to 5, wherein the duct has a height and depth dimension, said waveguide being disposed from the end of said duct a distance not less than three times the larger one of said height and depth duct dimension to obtain optimum attenuation.

7. An apparatus for the attenuation of sound from a source propagating in a duct comprising:

a microphone array disposed within said duct for detecting said source sound, said microphone array producing electrical signals representing the amplitude and phase characterishics of said source sound;

a speaker for generating cancelling sound of corresponding amplitude and 180° out of phase with said source sound, said speaker being disposed at a remote location from said duct;

an acoustic mixer connected to said duct and forming a continuation of said duct for the propagation of said source sound therethrough, said acoustic mixer being connected to said duct at a location spaced from said microphone array in the direction of propagation of said source sound;

a waveguide connecting said speaker with said acoustic mixer, said waveguide providing a path for the propagation of said cancelling sound from said speaker to said acoustic mixer for attenuation of said source sound;

an error microphone disposed within said acoustic mixer at a location spaced from said waveguide in the direction of propagation of said source sound, said error microphone sensing the acoustic output from said acoustic mixer and producing electrical signals representing the amplitude and phase characteristics of said acoustic output; and

electronic controller means connected with said microphone array, speaker and said error microphone, said electronic controller means being operable to delay, filter and scale the electrical signals from said microphone array to produce outputs for driving said speaker for generating said cancelling sound, and to deterministically adjust said outputs based on the electrical signals from said error microphone.

8. The apparatus of claim 7, wherein the duct has a height and depth dimension, said waveguide being disposed from the end of said duct a distance not less than three times the larger one of said height and depth duct dimensions to obtain optimum attenuation.

9. An apparatus for the attenuation of sound from at least one source propagating in a first and second duct, comprising:

first and second sensing means disposed in said first and second ducts respectively for sensing said source sound;

first and second cancelling means for generating cancelling sound of correponding amplitude and 180° out of phase with said source sound, said first and second cancelling means being disposed at a remote location from said first and second ducts;

first waveguide means connecting said first cancelling means with said first duct and second waveguide means connecting said second cancelling means with said second duct, said waveguide means being connected to said ducts at a location spaced from said sensing means in the direction of propagation of said source sound, said waveguide means providing a path for the propagation of said cancelling sound from said cancelling means to said ducts: and

an electronic controller means connected with said first and second sensing means and being operable to separately control and actuate said first and second cancelling means for the production of said cancelling sound to attenuate said source sound within said ducts and everywhere in the far field.

10. An apparatus for the attenuation of sound from at least one source propagating in a first and second duct comprising:

a first and second microphone array disposed within said first and second ducts respectively for detecting said source sound, said first and second microphone arrays producing electrical signals representing the amplitude and phase characteristics of said source sound;

a first and a second speaker for generating cancelling sound of corresponding amplitude and 180° out of phase with said source sound, said first and second speakers being disposed at a remote location from said first and second ducts;

a first and second acoustic mixer connected to said first and second ducts respectively and forming a continuation of said ducts for the propagation of said source sound therethrough, said

acoustic mixers being connected to said ducts at a location spaced from said microphone arrays in the direction of propagation of said source sound;

a first waveguide connecting said first speaker with said first acoustic mixer and a second waveguide connecting said second speaker with said second acoustic mixer, said waveguides providing a path for the propagation of said cancelling sound from said speakers to said acoustic mixers for attenuation of said source sound;

first and second error microphones disposed within said first and second acoustic mixers respectively at a location spaced from said waveguides in the direction of propagation of said source sound, said error microphones sensing the acoustic output from said acoustic mixers and providing electrical signals representing the amplitude and phase characteristics of said acoustic output;

an electronic controller means connected with said first and second microphone arrays, speakers and error microphones, said electronic controller means being operable to separately delay, filter and scale the electrical signals from said first and second microphone arrays to produce separate outputs for driving said first and second speakers for generating said cancelling sound, and to deterministically adjust said separate outputs based on the electrical signals received from said first and second error microphones respectively.

11. An apparatus for the attenuation of sound from a given source propagating in a duct comprising:

first sensing means disposed in said duct for detecting said source sound;

first cancelling means for generating first cancelling sound of corresponding amplitude and 180° out of phase with said source sound, said first cancelling means being disposed at a remote location from said duct;

first waveguide means connecting said first cancelling means with said duct, said first waveguide means being connected to said duct at a location spaced from said first sensing means in the direction of propagation of said source sound, said first waveguide means providing a path for the propagation of said first cancelling sound from said first cancelling means to said duct;

second sensing means disposed in said duct at a location spaced from said first waveguide means in the direction of propagation of said source sound, said second sensing means detecting said source sound attenuated by said first cancelling sound;

second cancelling means for generating second cancelling sound of corresponding amplitude and 180° out of phase with said source sound attenuated by said first cancelling sound, said second cancelling means being disposed at a remote location from said duct;

second waveguide means connecting said second cancelling means with said duct, said second waveguide means being connected to said duct at a location spaced from said second sensing means in the direction of propagation of said source sound, said second waveguide means providing a path for the propagation of said second cancelling sound from said second cancelling means to said duct; and

an electronic controller means connected with said first and second sensing means and being operable to separately actuate and control said first and second cancelling means for the production of said first cancelling sound and then said second cancelling sound to attenuate said source sound within said duct and everywhere in the far field.

12. An apparatus for the attenuation of sound from a source propagating in a duct comprising:

a first microphone array disposed within said duct for detecting said source sound, said first microphone array producing electrical signals representing the amplitude and phase characteristics of said source sound;

a first speaker for generating first cancelling sound of corresponding amplitude and 180° out of phase with said source sound, said first speaker being disposed at a remote location from said duct;

a first acoustic mixer connected to said duct and forming a continuation of said duct for the propagation of said source sound therethrough, said first acoustic mixer being connected to said duct at a location spaced from said first microphone array in the direction of propagation of said source sound;

a first waveguide connecting said first speaker with said first acoustic mixer, said first waveguide providing a path for the propagation of said first cancelling sound from said first speaker to said first acoustic mixer for attenuation of said source sound;

a first error microphone disposed within said first acoustic mixer at a location spaced from said first waveguide in the direction of propagation of said source sound, said first error microphone sensing the acoustic output from said first acoustic mixer and producing electrical signals representing the amplitude and phase characteristics of said acoustic output;

a second microphone array disposed within said duct at a location spaced from said first error microphone in the direction of propagation of said source sound for detecting said source sound attenuated by said first cancelling sound, said second microphone array producing electrical signals representing the amplitude and phase characteristics of said attenuated source sound;

a second speaker for generating second cancelling sound of corresponding amplitude and 180° out of phase with said attenuated source sound, said second speaker being disposed at a remote location from said duct;

a second acoustic mixer connected to said duct and forming a continuation of said duct for the propagation of said attenuated source sound therethrough, said second acoustic mixer being connected to said duct at a location spaced from said second microphone array in the direction of propagation of said attenuated source sound;

a second waveguide connecting said second speaker with said second acoustic mixer, said second waveguide providing a path for the propagation of said second cancelling sound from said speaker to said second acoustic mixer for further attenuation of said attenuated source sound;

a second error microphone disposed within said second acoustic mixer at a location spaced from said second waveguide in the direction of propagation of said attenuated source sound, said second error microphone sensing the acoustic output of said second acoustic mixer and producing electrical signals representing the amplitude and phase characteristics of said acoustic output; and

an electronic controller means connected with said first and second microphone arrays, speakers and error microphones, said electronic controller means being operable to separately delay, filter and scale the electrical signals from said first and second microphone arrays to produce a first and second output to drive said first and second speakers respectively for the production of said first and second cancelling sound, and to deterministically adjust said outputs based on the electrical signals from said first and second error microphones respectively.

13. An apparatus for the attenuation of sound from a given source propagating in a duct having height and depth dimensions comprising:

sensing means for detecting said source sound;

cancelling means for generating cancelling sound of corresponding amplitude and 180° out of phase with said source sound, said cancelling means being disposed at a remote location from said duct;

waveguide means connecting said cancelling means with said duct, said waveguide means being connected to said duct at a location spaced from said sensing means in the direction of propagation of said source sound, and being spaced at least three times the larger one of said height and depth duct dimensions from the end of said duct, said waveguide means providing a path for the propagation of said cancelling sound from said cancelling means to said duct; and

electronic controller means connected with said sensing means and being operable to actuate and control said cancelling means for the production of said cancelling sound to attenuate said source sound within said duct and everywhere in the far field.

14. An apparatus for the attenuation of sound from a source propagating in a duct having height and depth dimensions comprising:

a microphone array disposed within said duct for detecting said source sound, said microphone array producing electrical signals representing the amplitude and phase characteristics of said source sound;

a speaker for generating cancelling sound of corresponding amplitude and 180° out of phase with said source sound, said speaker being disposed at a remote location from said duct;

an acoustic mixer connected to said duct and forming a continuation of said duct for the propagation of said source sound therethrough, said acoustic mixer being connected to said duct at a location spaced from said microphone array in the direction of propagation of said source sound;

a waveguide connecting said speaker with said acoustic mixer and being spaced at least three times the larger one of said height and depth duct dimensions from the end of said duct, said waveguide providing a path for the propagation of said cancelling sound from said speaker to said acoustic mixer for attenuation of said source sound;

an error microphone disposed within said acoustic mixer at a location spaced from said waveguide in the direction of propagation of said source sound, said error microphone sensing the acoustic output from said acoustic mixer and providing electrical signals representing the amplitude and phase characteristics of said acoustic output; and

electronic controller means connected with said microphone array, speaker and said error microphone, said electronic controller means being operable to delay, filter and scale the electrical signals from said microphone array to produce outputs for driving said speaker for the production of said cancelling sound, and to deterministically adjust said outputs based on the electrical signals from said error microphone.

15. An apparatus for the attenuation of sound from a given source propagating in a duct comprising:

sensing means for detecting said source sound;

cancelling means for generating cancelling sound of corresponding amplitude and 180° out of phase with said source sound, said cancelling means being disposed at a remote location from said duct;

a pair of waveguide means connecting said cancelling means with said duct, said waveguide means being connected parallel to and on opposite sides of said duct at a location spaced from said sensing means in the direction of propagation of said source sound, said waveguide means introducing said cancelling sound into said duct in the direction of propagation of said source sound thereby preventing the first cross mode of said source sound from developing in said duct;

electronic controller means connected with said sensing means and being operable to actuate and control said cancelling means for the production of said cancelling sound to attenuate said source sound within said duct and everywhere in the far field.

16. An apparatus for the attenuation of sound from a source propagating in a duct comprising:

a microphone array disposed within said duct for detecting said source sound, said microphone array producing electrical signals representing the amplitude and phase characteristics of said source sound;

a speaker for generating cancelling sound of corresponding amplitude and 180° out of phase with said source sound, said speaker being disposed at a remote location from said duct;

an acoustic mixer connected to said duct and forming a continuation of said duct for the propagation of said source sound therethrough, said acoustic mixer being connected to said duct at a location spaced from said microphone array in the direction of propagation of said source sound;

a pair of waveguides for connecting said speaker to said acoustic mixer, said waveguides being attached perpendicular to said acoustic mixer and in alignment with one another on opposite sides of said duct to provide separate paths for the propagation of said cancelling sound from said speaker to said acoustic mixer and thereby preventing the first cross mode of said source sound from developing within said duct;

an error microphone disposed within said acoustic mixer at a location spaced from said waveguide in the direction of propagation of said source sound, said error microphone sensing the acoustic output from said acoustic mixer and producing electrical signals representing the amplitude and phase characteristics of said output; and

electronic controller means connected with said microphone array, speaker and said error microphone, said electronic controller means being operable to delay, filter and scale the electrical signals from said microphone array to produce outputs for driving said speaker for the production of said cancelling sound, and to deterministically adjust said outputs based on the electrical signal from said error microphone.

17. An apparatus for the attenuation of sound from a source propagating in a duct comprising:

a microphone array disposed within said duct for detecting said source sound, said microphone array producing electrical signals representing the amplitude and phase characteristics of said source sound;

a speaker for generating cancelling sound, said speaker being disposed at a remote location from said duct;

an acoustic mixer connected to said duct and forming a continuation of said duct for the propagation of said source sound therethrough, said acoustic mixer being connected to said duct at a location spaced from said microphone array in the direction of propagation of said source sound;

a waveguide connecting said speaker with said acoustic mixer, said waveguide providing a path for the propagation of said cancelling sound from said speaker to said acoustic mixer for attenuation of said source sound;

an error microphone disposed within said acoustic mixer at a location spaced from said waveguide in the direction of propagation of said source sound, said error microphone sensing the acoustic output from said acoustic mixer and producing electrical signals representing the amplitude and phase characteristics of said acoustic output; and

electronic controller means including an adaptive cancelling filter, an amplifier, a phase correction filter and DC loop means, said adaptive cancelling filter being operable to delay, filter and scale said electrical signals from said microphone array and produce outputs to said amplifier, said amplifier driving said speaker to produce said cancelling sound, said phase compensation filter adjusting the phase characteristics of said electrical signal produced by said error microphone and said DC loop means providing a DC signal for introduction into said adaptive cancelling filter to assure stability of the operation thereof, said adaptive cancelling filter being operable to deterministically adjust said delay, filtering and scaling of said electrical signals from said microphone array based on said electrical signals received from said error microphone to produce outputs to drive said speaker for the production of cancelling sound having the mirror image amplitude and phase characteristics of said source sound within preset limits.

18. The apparatus of claim 17, wherein said adaptive filter includes a transversal filter and a modified LMS algorithm governing the operation of said transversal filter.

19. The apparatus of claim 18, wherein said LMS algorithm is modified to accommodate the acoustic delays of said apparatus, said acoustic delays including the time required for said source sound to propagate from said microphone array to said error microphone, and the time required for said cancelling sound to propagate from said speaker to said acoustic mixer.

20. An apparatus for the attenuation of sound from at least one source propagating in a first and second duct comprising:

a first and second microphone array disposed within said first and second ducts respectively for detecting said source sound, said first and second microphone arrays producing electrical signals representing the amplitude and phase characteristics of said source sound;

a first and second speaker for generating cancelling sound, said first and second speakers being disposed at a remote location from said first and second ducts;

a first and second acoustic mixer connected to said first and second ducts respectively and forming a continuation of said ducts for the propagation of said source sound therethrough, said acoustic mixers being connected to said ducts at a location spaced from said microphone arrays in the direction of propagation of said source sound;

a first waveguide connecting said first speaker with said first acoustic mixer and a second waveguide connecting said second speaker with said second acoustic mixer, said waveguides providing a path for the propagation of said cancelling sound from said speakers to said acoustic mixers for attenuation of said source sound;

first and second error microphones disposed within said first and second acoustic mixers respectively at a location spaced from said waveguides in the direction of propagation of said source sound, said error microphones sensing the acoustic outputs from said acoustic mixers and providing electrical signals representing the amplitude and phase characteristics of said acoustic outputs; and

electronic controller means including an adaptive cancelling filter, an amplifier, a phase correction filter and DC loop means, said adaptive cancelling filter being operable to separately delay, filter and scale said electrical signals from said first and second microphone arrays and produce separate outputs to said amplifier for driving said first and second speakers to produce said cancelling sound, said phase compensation filter adjusting the phase characteristics of said electrical signals produced by said first and second error microphones and said DC loop means providing a DC signal for introduction into said adaptive cancelling filter to assure stable operation thereof, said adaptive cancelling filter being operable to deterministically adjust said delay, filtering and scaling of said electrical signals from said first and second microphone arrays based on said electrical signals received from said first and second error microphones respectively to produce separate outputs to drive said first and second speakers for the production of cancelling sound having the mirror image amplitude and phase characteristics of said source sound within preset limits.

21. An apparatus for the attenuation of sound from a source propagating in a duct comprising:

a first microphone array disposed within said duct for detecting said source sound, said first microphone array producing electrical signals representing the amplitude and phase characteristics of said source sound;

a first speaker for generating first cancelling sound, said first speaker being disposed at a remote location from said duct;

a first acoustic mixer connected to said duct and forming a continuation of said duct for the propagation of said source sound therethrough, said first acoustic mixer being connected to said duct at a location spaced from said first microphone array in the direction of propagation of said source sound;

a first waveguide connecting said first speaker with said first acoustic mixer, said first waveguide providing a path for the propagation of said first cancelling sound from said first speaker to said first acoustic mixer for attenuation of said source sound;

a first error microphone disposed within said first acoustic mixer at a location spaced from said first waveguide in the direction of propagation of said source sound, said error microphone sensing the acoustic output from said first acoustic mixer and producing electrical signals, representing the amplitude and phase characteristics of said acoustic output;

a second microphone array disposed within said duct at a location spaced from said first error microphone in the direction of propagation of said source sound for detecting said source sound attenuated by said first cancelling sound, said second microphone array producing electrical signals representing the amplitude and phase characteristics of said attenuated source sound;

a second speaker for generating second cancelling sound, said second speaker being disposed at a remote location from said duct;

a second acoustic mixer connected to said duct and forming a continuation of said duct for the propagation of said attenuated source sound therethrough, said second acoustic mixer being connected to said duct at a location spaced from said second microphone array in the direction of propagation of said attenuated source sound;

a second waveguide connecting said second speaker with said second acoustic mixer, said second waveguide providing a path for the propagation of said second cancelling sound from said speaker to said second acoustic mixer for further attenuation of said attenuated source sound;

a second error microphone disposed within said second acoustic mixer at a location spaced from said second waveguide in the direction of propagation of said attenuated source sound, said second error microphone sensing the acoustic output from said second acoustic mixer and producing electrical signals representing the amplitude and phase characteristics of said acoustic output; and

electronic controller means including an adaptive cancelling filter, an amplifier, a phase correction filter and DC loop means, said adaptive cancelling filter being operable to separately delay, filter and scale said electrical signals from said first and second microphone arrays and produce separate output to said amplifier for driving said first and second speakers to produce said first and second cancelling sound respectively, said phase compensation filter adjusting the phase characteristics of said electrical

signals produced by said first and second error microphones and said DC loop providing a DC signal for introduction into said adaptive cancelling filter to assure stable operation thereof, said adaptive cancelling filter being operable to deterministically adjust said delay, filtering and scaling of said electrical signals from said first and second microphone arrays based on said electrical signals received from said first and second error microphones respectively to produce separate outputs to drive said first and second speakers for the production of said first and second cancelling sounds having the mirror image amplitude and phase characteristics of said source sound and attenuated source sound within preset limits.

22. An apparatus for the attenuation of sound from a given source propagating in a duct comprising:

a microphone array disposed within said duct for detecting said source sound; said microphone array producing electrical signals representing the amplitude and phase characteristics of said source sound:

a speaker for generating cancelling sound, said speaker being disposed at a remote location from said duct;

a pair of waveguides connecting said speaker with said duct, said waveguides being connected parallel to and on opposite sides of said duct at a location spaced from said microphone array in the direction of propagation of said source sound, said waveguides introducing said cancelling sound into said duct in the direction of propagation of said source sound thereby preventing the first cross mode of said source sound from developing in said duct;

an error microphone disposed within said duct at a location spaced from said waveguides in the direction of propagation of said source sound, said error microphone sensing the acoustic output from said duct and providing electrical signals representing the amplitude and phase characteristics of said acoustic output; and

electronic controller means including an adaptive cancelling filter, an amplifier, a phase correction filter and DC loop means, said adaptive cancelling filter being operable to delay, filter and scale said electrical signals from said microphone array and produce outputs to said amplifier for driving said speaker to produce said cancelling sound, said phase compensation filter adjusting the phase characteristics of said electrical signals produced by said error microphone and said DC loop providing a DC signal for introduction into said adaptive cancelling filter to assure stable operation thereof, said adaptive cancelling filter being operable to deterministically adjust said delay, filtering and scaling of said electrical signals from said microphone array based on said electrical signals received from said error microphone to produce outputs to drive said speaker for the production of said cancelling sound having the mirror image amplitude and phase characteristics of said source sound within preset limits.

23. An apparatus for the attenuation of sound from a source propagating in a duct comprising:

a microphone array disposed within said duct for detecting said source sound, said microphone array producing electrical signals representing the amplitude and phase characteristics of said source sound;

a speaker for generating cancelling sound, said speaker being disposed at a remote location from said duct;

an acoustic mixer connected to said duct and forming a continuation of said duct for the propagation of said source sound therethrough, said acoustic mixer being connected to said duct at a location spaced from said microphone array in the direction of propagation of said source sound;

a pair of waveguides for connecting said speaker to said acoustic mixer, said waveguides being attached perpendicular to said duct and in alignment with one another on opposite sides of said duct to provide separate paths for the propagation of said cancelling sound from said speaker to said acoustic mixer and thereby preventing the first cross mode of said source sound from developing within said duct;

an error microphone disposed within said acoustic mixer at a location spaced from said waveguide in the direction of propagation of said source sound, said error microphone sensing the acoustic output from said acoustic mixer and producing electrical signals representing the amplitude and phase characteristics of said acoustic output; and

electronic controller means including an adaptive cancelling filter, an amplifier, a phase correction filter and DC loop means, said adaptive cancelling filter being operable to delay, filter and scale said electrical signals from said microphone array and produce outputs to said amplifier for driving said speaker to produce said cancelling sound, said phase compensation filter adjusting the phase characteristics of said electrical signals produced by said error microphone and said DC loop providing a DC signal for introduction into said adaptive cancelling filters to assure stable operation thereof, said adaptive cancelling filter being operable to deterministically adjust said delay, filtering and scaling of said electrical signals from said microphone array based on said electrical signals received from said error microphone to produce revised outputs to drive said speaker for the production of said cancelling sound having the mirror image amplitude and phase characteristics of said source sound within preset limits.

24. An apparatus for the attenuation of sound from a source propagating in a duct comprising:

a microphone array disposed within said duct for detecting said source sound, said microphone

array producing electrical signals representing the amplitude and phase characteristics of sound within said duct;

a speaker for generating cancelling sound, said speaker being disposed at a remote location from said duct;

an acoustic mixer connected to said duct and forming a continuation of said duct for the propagation of source sound therethrough, said acoustic mixer being connected to said duct at a location spaced from said microphone array in the direction of propagation of said source sound;

a waveguide connecting said speaker with said acoustic mixer, said waveguide providing a path for the propagation of said cancelling sound from said speaker to said acoustic mixer for attenuation of said source sound, said cancelling sound producing at least some standing waves propagating in the opposite direction of said source sound, said standing waves being detectable by said microphone array and being incorporated into said electrical signal representing the amplitude and phase characteristics of sound within said duct;

an error microphone disposed within said acoustic mixer at a location spaced from said waveguide in the direction of propagation of said source sound, said error microphone sensing the acoustic output of said acoustic mixer and producing electrical signals representing the amplitude and phase characteristics of said acoustic output; and

electronic circuitry means including an adaptive cancelling filter, an adaptive uncoupling filter, an adaptive compensation filter, DC loop means for introducing a DC signal to said adaptive cancelling filter, and low pass filter means for limiting the frequency range of signals within said electronic circuitry, said adaptive uncoupling filter being operable to substantially remove the component of said electrical signal produced by said microphone array representing said standing waves generated by said cancelling sound and sensed by said microphone array, said adaptive compensation filter being operable to deterministically adjust the phase characteristics of said electrical signal produced by said error microphone for introduction into said adaptive cancelling filter to assure stable operation thereof, said adaptive cancelling filter being operable to successively delay, filter and scale said electrical signals from said microphone array to produce successive outputs for driving said speaker, and then to deterministically adjust said delay, filtering and scaling operating based on said phase compensated electrical signals received from said error microphone to provide successive revised outputs for driving said speaker to produce cancelling sound having the mirror image amplitude and phase characteristics of said source sound within preset limits.

25. The apparatus of claim 24, wherein said low pass filters provide an inherent processing delay in the function of said electronic circuitry, and inherent acoustic delays being produced by the time of propagation of said source sound from said microphone array to said error microphone, and the time of propagation of said cancelling sound from said speaker to said acoustic mixer; whereby to accommodate said delays the physical spacing between said microphone array, said speaker, and said error microphone must be in accordance with the following relationship:

$$T>\lambda min/4+d_1+d_2+D_f$$

wherein

T=spacing between microphone array and error microphone

$\lambda$min=shortest wavelength of interest in the source sound

$d_1$=spacing between the speaker and acoustic mixer

$d_2$=spacing between the waveguide and error microphone

$D_f$=delay associated with the low pass filters.

26. An apparatus for the attenuation of sound from at least one source propagating in a first and second duct comprising:

a first and second microphone array disposed within said first and second ducts respectively for detecting said source sound, said first and second microphone arrays producing electrical signals representing the amplitude and phase characteristics of said sound within said duct;

a first and second speaker for generating cancelling sound, said first and second speakers being disposed at a remote location from said first and second ducts;

a first and second acoustic mixer connected to said first and second ducts respectively and forming a continuation of said ducts for the propagation of said source sound therethrough, said acoustic mixers being connected to said ducts at a location spaced from said microphone array in the direction of propagation of said source sound;

a first waveguide connecting said first speaker with said first acoustic mixer and a second waveguide connecting said second speaker with said second acoustic mixer, said waveguides providing a path for the propagation of said cancelling sound from said speakers to said acoustic mixers for the attenuation of said source sound, said cancelling sound producing at least some standing waves propagating in the opposite direction of said source sound, said standing waves being detectable by said microphone array and being incorporated as a component of said electrical signals produced by said microphone array;

first and second error microphones disposed within said first and second acoustic mixers

respectively at a location spaced from said waveguides in the direction of propagation of said source sound, said error microphones sensing the acoustic output from said acoustic mixers and providing electrical signals representing the amplitude and phase characteristics of said acoustic outputs;

electronic circuitry means including an adaptive cancelling filter, an adaptive uncoupling filter, an adaptive compensation filter, DC loop means for introducing a DC signal to said adaptive cancelling filter, and low pass filter means for limiting the frequency range of signals within said electronic circuitry, said adaptive uncoupling filter being operable to substantially remove the component of said electrical signals produced by said first and second microphone arrays representing said standing waves generated by said cancelling sound and sensed by said microphone arrays, said adaptive compensation filter being operable to deterministically adjust the phase characteristics of said electrical signals produced by said first and second error microphones for introduction into said adaptive cancelling filter to assure stable operation thereof, said adaptive cancelling filter being operable to successively delay, filter and scale said electrical signals from each of said first and second microphone arrays to produce separate, successive outputs for driving said first and second speakers, and then to deterministically adjust said delay, filtering and scaling operation based on said phase compensated electrical signals received from said first and second error microphones to provide separate, successive revised outputs for driving said first and second speakers to produce cancelling sound having the mirror image amplitude and phase characteristics of said source sound within preset limits.

27. The apparatus of claim 26, wherein said low pass filters provide an inherent processing delay in the function of said electronic circuitry, and inherent acoustic delays being produced by the time of propagation of said source sound from said microphone array to said error microphone, and the time of propagation of said cancelling sound from said speaker to said acoustic mixer; whereby to accommodate said delays the physical spacing between said microphone array, said speaker, and said error microphone must be in accordance with the following relationship:

$$T>\lambda min/4+\delta_1+d_2+D_f$$

wherein

$T$=spacing between microphone array and error microphone

$\lambda min$=shortest wavelength of interest in the source sound

$d_1$=spacing between the speaker and acoustic mixer

$d_2$=spacing between the waveguide and error microphone

$D_f$=delay associated with the low pass filters.

28. An apparatus for the attenuation of sound from a source propagating in a duct comprising:

a first microphone array disposed within said duct for detecting said source sound, said first microphone array producing electrical signals representing the amplitude and phase characteristics of said sound, within said duct;

a first speaker for generating first cancelling sound, said first speaker being disposed at a remote location from said duct;

a first acoustic mixer connected to said duct and forming a continuation of said duct for the propagation of said source sound therethrough, said first acoustic mixer being connected to said duct at a location spaced from said first microphone array in the direction of propagation of said source sound;

a first waveguide connecting said first speaker with said first acoustic mixer, said first waveguide providing a path for the propagation of said first cancelling sound from said first speaker to said first acoustic mixer for attenuation of said source sound, said first cancelling sound producing at least some standing waves propagating in the opposite direction of said source sound, said standing waves being detectable by said first microphone array and being incorporated as a component of said electrical signals produced by said first microphone array;

a first error microphone disposed within said first acoustic mixer at a location spaced from said first waveguide in the direction of propagation of said source sound, said error microphone sensing the acoustic output from said first acoustic mixer and producing electrical signals representing the amplitude and phase characteristics of said acoustic output;

a second microphone array disposed within said duct at a location spaced from said first error microphone in the direction of propagation of said source sound for detecting said source sound attenuated by said first cancelling sound, said second microphone array producing electrical signals representing the amplitude and phase characteristics of said attenuated source sound;

a second speaker for generating second cancelling sound, said second speaker being disposed at a remote location from said duct;

a second acoustic mixer connected to said duct and forming a continuation of said duct for the propagation of said attenuated source sound therethrough, said second acoustic mixer being connected to said duct at a location spaced from said second microphone array in the direction of propagation of said attenuated source sound;

a second waveguide connecting said second speaker with said second acoustic mixer, said second waveguide providing a path for the propagation of said second cancelling sound from said

speaker to said second acoustic mixer for further attenuation of said attenuated source sound, said second cancelling sound producing at least some standing waves propagating in the opposite direction of said source sound, said standing waves being detectable by said second microphone array and being incorporated as a component of said electrical signals produced by said second microphone array;

a second error microphone disposed within said second acoustic mixer at a location spaced from said second waveguide in the direction of propagation of said attenuated source sound, said second error microphone sensing the acoustic output from said second acoustic mixer and producing electrical signals representing the amplitude and phase characteristics of said output; and

electronic circuitry means including an adaptive cancelling filter, an adaptive uncoupling filter, an adaptive compensation filter, DC loop means for introducing a DC signal to said adaptive cancelling filter, and low pass filter means for limiting the frequency range of signals within said electronic circuitry, said adaptive uncoupling filter being operable to substantially remove the components of said electrical signals produced by said first and second microphone arrays representing said standing waves generated by said first and second cancelling sounds and sensed by said first and second microphone arrays, said adaptive compensation filter being operable to deterministically adjust the phase characteristics of said electrical signals produced by said first and second error microphones for introduction into said adaptive cancelling filter to assure stable operation thereof, said adaptive cancelling filter being operable to successively delay, filter and scale said electrical signals from said first and second microphone arrays to produce separate successive outputs for driving said first and second speakers, and then to deterministically adjust said delay, filtering and scaling operation based on said phase compensated electrical signals received from said first and second error microphones to provide separate successive revised outputs for driving said first and second speakers to produce first and second cancelling sounds having the mirror image amplitude and phase characteristics of said source sound within preset limits.

29. An apparatus for the attenuation of sound from a source propagating in a duct comprising:

a microphone array disposed within said duct for detecting said source sound, said microphone array producing electrical signals representing the amplitude and phase characteristics of said source sound;

a speaker for generating cancelling sound, said speaker being disposed at a remote location from said duct;

a pair of waveguides connecting said speaker with said duct, said waveguides being connected parallel to and on opposite sides of said duct at a location spaced from said microphone array in the direction of propagation of said source sound, said waveguides introducing said cancelling sound into said duct in parallel relation to the direction of propagation of said source sound thereby preventing the first cross mode of said source sound from developing in said duct, said cancelling sound producing at least some standing waves propagating in the opposite direction of said source sound, said standing waves being detectable by said microphone array and being incorporated as a component of said electrical signals produced by said microphone array;

an error microphone disposed within said duct at a location spaced from said waveguides in the direction of propagation of said source sound, said error microphone sensing the acoustic output from said duct and producing electrical signals representing the amplitude and phase characteristics of said acoustic output; and

electronic circuitry means including an adaptive cancelling filter, an adaptive uncoupling filter, an adaptive compensation filter, DC loop means for introducing a DC signal to said adaptive cancelling filter, and low pass filter means for limiting the frequency range of signals within said electronic circuitry, said adaptive uncoupling filter being operable to substantially remove the component of said electrical signal produced by said microphone array representing said standing waves generated by said cancelling sound and sensed by said microphone array, said adaptive compensation filter being operable to deterministically adjust the phase characteristics of said electrical signal produced by said error microphone for introduction into said adaptive cancelling filter to assure stable operation thereof, said adaptive cancelling filter being operable to successively delay, filter and scale said electrical signals from said microphone array to produce successive outputs for driving said speaker, and then to deterministically adjust said delay, filtering and scaling operation based on said phase compensated electrical signals received from said error microphone to provide successive revised outputs for driving said speaker to produce cancelling sound having the mirror image amplitude and phase characteristics of said source sound within preset limits.

30. An apparatus for the attenuation of sound from a source propagating in a duct comprising:

a microphone array disposed within said duct for detecting said source sound, said microphone array producing electrical signals representing the amplitude and phase characteristics of said source sound;

a speaker for generating cancelling sound, said speaker being disposed at a remote location from said duct;

an acoustic mixer connected to said duct and forming a continuation of said duct for the propagation of said source sound therethrough, said acoustic mixer being connected to said duct at a location spaced from said microphone array in the direction of propagation of said source sound;

a pair of waveguides for connecting said speaker to said acoustic mixer, said waveguides being attached perpendicular to said duct and in alignment with one another on opposite sides of said duct to provide separate paths for the propagation of said cancelling sound from said speaker to said acoustic mixer and thereby preventing the first cross mode of said source sound from developing within said duct, said cancelling sound producing at least some standing waves propagating in the opposite direction of said source sound, said standing waves being detectable by said microphone array and being incorporated as a component of said electrical signals produced by said microphone array;

an error microphone disposed within said acoustic mixer at a location spaced from said waveguide in the direction of propagation of said source sound, said error microphone sensing the acoustic output from said acoustic mixer and producing electrical signals representing the amplitude and phase characteristics of said acoustic output;

electronic circuitry means including an adaptive cancelling filter, an adaptive uncoupling filter, an adaptive compensation filter, DC loop means for introducing a DC signal to said adaptive cancelling filter, and low pass filter means for limiting the frequency range of signals within said electronic circuitry, said adaptive uncoupling filter being operable to substantially remove the component of said electrical signal produced by said microphone array representing said standing waves generated by said cancelling sound and sensed by said microphone array, said adaptive compensation filter being operable to deterministically adjust the phase characteristics of said electrical signal produced by said error microphone for introduction into said adaptive cancelling filter to assure stable operation thereof, said adaptive cancelling filter being operable to successively delay, filter and scale said electrical signals from said microphone array to produce successive outputs for driving said speaker, and then to deterministically adjust said delay, filtering and scaling operation based on said phase compensated electrical signals received from said error microphone to provide successive revised outputs for driving said speaker to produce cancelling sound having the mirror image amplitude and phase characteristics of said source sound within preset limits.

31. An apparatus for the cancellation of vibration in a physical system comprising:

first sensing means for detecting said vibration, said first sensing means producing first electrical signals representing the amplitude and phase characteristics of said vibration;

cancelling means for generating cancelling vibration for combination with said vibration from said physical system;

second sensing means disposed at a location spaced from said first sensing means in the direction of propagation of said vibration, said second sensing means detecting the summation of said vibration from said physical system and said cancelling vibration and producing second electrical signals representing the amplitude and phase characteristics of said summation; and

an adaptive filter, said adaptive filter including a modified LMS algorithm and a transversal filter, said adaptive filter being operable to successively delay, filter and scale said first electrical signals from said first sensing means to produce corresponding outputs for driving said cancelling means, and then to deterministically adjust said delay, filtering and scaling operation based on said second electrical signals received from said second sensing means to provide corresponding revised outputs for driving said cancelling means to produce cancelling vibration having the mirror image amplitude and phase characteristics of said vibration from said physical system within preset limits.

32. The apparatus of claim 31, wherein said modified LMS algorithm includes means for accommodating the delays inherent in the propagation of said vibration from said first sensing means to said second sensing means, and the propagation of said cancelling vibration from said cancelling means to the point of combination of said cancelling vibration with said vibration from said physical system.

33. An apparatus for the cencellation of vibration in a physical system comprising:

first sensing means for detecting said vibration, said first sensing means producing first electrical signals representing the amplitude and phase characteristics of said vibration;

cancelling means for generating cancelling vibration for combination with said vibration from said physical system;

second sensing means disposed at a location spaced from said first sensing means in the direction of propagation of said vibration, said second sensing means detecting the summation of said vibration from said physical system and said cancelling vibration and producing second electrical signals representing the amplitude and phase characteristics of said summation; and

an adaptive filter for driving said cancelling means to produce said cancelling vibration, said adaptive filter including a transversal filter and a LMS algorithm modified to accommodate the delay inherent in the propagation of said vibration from said first sensing means to said second sensing means, and the delay inherent in the propagation of said cancelling vibration from said cancelling means to the point of combination of said cancelling vibration with said vibration from said physical system, said adaptive filter being operable to successively delay, filter and scale said first electrical signals from said first sensing means to produce corresponding outputs for driving said cancelling means, and then to deterministically adjust said delay, filtering and scaling operation based on said second electrical signals received from said second sensing means to provide corresponding revised

outputs for driving said cancelling means to produce cancelling vibration having the mirror image amplitude and phase characteristics of said vibration from said physical system within preset limits.

34. An apparatus for the cancellation of vibration in a physical system comprising:

first sensing means for detecting said vibration, said first sensing means producing first electrical signals representing the amplitude and phase characteristics of said vibration;

cancelling means for generating cancelling vibration for combination with said vibration from said physical system;

second sensing means disposed at a location spaced from said first sensing means in the direction of propagation of said vibration, said second sensing means detecting the summation of said vibration from said physical system and said cancelling vibration and producing second electrical signals representing the amplitude and phase characteristics of said summation; and

electronic circuitry means including an adaptive cancelling filter and an adaptive compensation filter, said adaptive compensation filter being operable to deterministically adjust the phase characteristics of said second electrical signals produced by said second sensing means for introduction into said adaptive cancelling filter to assure stable operation thereof, said adaptive cancelling filter having a modified LMS algorithm and a transversal filter, said adaptive filter being operable to successively delay, filter and scale said first electrical signals from said first sensing means to produce corresponding outputs for driving said cancelling means, and then to deterministically adjust said delay, filtering and scaling operation based on said phase compensated second electrical signals received from said second sensing means to provide corresponding revised outputs for driving said cancelling means to produce cancelling vibration having the mirror image amplitude and phase characteristics of said vibration from said physical system within preset limits.

35. The apparatus of claim 34, wherein said modified LMS algorithm, includes means for accommodating the delays inherent in the propagation of said vibration from said first sensing means to said second sensing means, and the propagation of said cancelling vibration from said cancelling means to the point of combination of said cancelling vibration with said vibration from said physical system.

36. An apparatus for the cancellation of vibration in a physical system comprising:

first sensing means for detecting said vibration, said first sensing means producing first electrical signals representing the amplitude and phase characteristics of said vibration;

cancelling means for generating cancelling vibration for combination with said vibration from said physical system;

second sensing means disposed at a location spaced from said first sensing means in the direction of propagation of said vibration, said second sensing means detecting the summation of said vibration from said physical system and said cancelling vibration and producing second electrical signals representing the amplitude and phase characteristics of said summation; and

electronic circuitry means including an adaptive cancelling filter for diving said cancelling means to produce said cancelling vibration, and an adaptive compensation means operable to deterministically adjust the phase characteristics of said second electrical signals produced by said second sensing means for introduction into said adaptive cancelling filter to assure stable operation thereof, said adaptive cancelling filter including a transversal filter and a LMS algorithm modified to accommodate the delay inherent in the propagation of said vibration from said first sensing means to said second sensing means, and the delay inherent in the propagation of said cancelling vibration from said cancelling means to the point of combination of said cancelling vibration with said vibration from said physical system, said adaptive filter being operable to successively delay, filter and scale said first electrical signals from said first sensing means to produce corresponding outputs for driving said cancelling means, and then to deterministically adjust said delay, filtering and scaling operation based on said second electrical signals received from said second sensing means to provide corresponding revised outputs for driving said cancelling means to produce cancelling vibration having the mirror image amplitude and phase characteristics of said vibration from said physical system within preset limits.

37. Apparatus for the attenuation of sound from a source propagating in a duct or for cancellation of vibration in a physical system and substantially as herein described with reference to Figures 5 to 16 of the accompanying drawings.

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TITLE: Acoustic attenuators with active sound cancelling - has adaptive filter which drives speaker for production of cancelling sound 180 degrees of out phase

INVENTOR: POOLE, L A; TICHY, J ; WARNAKA, G E

PATENT-ASSIGNEE: ACTIVE NOISE & VIBRATION TECHN[ACTIN] , LORD CORP[LORD]

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| GB **2088951** A | June 16, 1982 | N/A | 035 | N/A |
| CA 1161766 A | February 7, 1984 | N/A | 000 | N/A |
| DE 3144052 A | July 8, 1982 | N/A | 000 | N/A |
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| GB 2088951A | N/A | 1981GB-0032854 | October 30, 1981 |
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ABSTRACTED-PUB-NO: DE 3144052C

BASIC-ABSTRACT:

The microphone array (33) senses source sound in a duct (13) and converts it to an electrical signal which is sent to an electronic adaptive filter (23). The adaptive filter drives a speaker (17) for the production of cancelling sound 180 degrees out of phase with the source sound, which propagates through a waveguide (19) to an acoustic mixer (15). A microphone (35) located in a position downstream from the waveguide within the acoustic mixer produces a signal which represents the error between the attenuation achieved in the acoustic mixer and the desired attenuation based on preset levels.

This error signal is introduced into the adaptive filter (23) which then adjusts its signal driving the speaker so that the cancelling sound propagating into the acoustic mixer more nearly approx. the mirror image of the source sound. The adaptive filter has a modified LMS (least means square) algorithm governing the operation of its transversal filter which accommodates the

inherent delays associated with the propagation of acoustic waves in a duct. A number of wave guides may connect the speaker to the acoustic mixer. A single electronic circuit may provide the signal processing for a pair of ducts.

ABSTRACTED-PUB-NO: GB 2088951A

EQUIVALENT-ABSTRACTS:

An active noise damper has at least one input sensor (33) to detect the noise in a channel (13) from a source and generate a first signal representing the amplitude and phase characteristics of the noise. A canceller (17) generates cancelling noise which is fed into the channel (13) to compensate for the source noise. An error detector (35) detects the combined source and cancelling noise and generates a second signal representing the amplitude and phase of the noise after compensation. The second signal is delayed due to the transmission time from the sensor (33) to the canceller (17) and the error detector (35). A control is connected to the input sensor (33), the canceller (17) and the error detector (35) to activate and control the canceller (17). The control includes an adjustable filter (23) to adjust the delay.

A mixer (15) connected to the channel (13) at a distance from the input sensor (33) forms a continuation of the channel (13). A wave guide (19) connects the canceller (17) to the mixer (15) and provides a path for the canceller (17) to the mixer (15) and provides a path for the cancelling noise. The distance between the guide (19) and the end of the channel (13) is not less than three times the max. cable diameter.

ADVANTAGE - Simple construction and efficiently allows optimal damping.

GB 2088951B

The microphone array (33) senses source sound in a duct (13) and converts it to an electrical signal which is sent to an electronic adaptive filter (23). The adaptive filter drives a speaker (17) for the production of cancelling sound 180 degrees out of phase with the source sound, which propagates through a waveguide (19) to an acoustic mixer (15). A microphone (35) located in a position downstream from the waveguide within the acoustic mixer produces a signal which represents the error between the attenuation achieved in the acoustic mixer and the desired attenuation based on preset levels.

This error signal is introduced into the adaptive filter (23) which then adjusts its signal driving the speaker so that the cancelling sound propagating into the acoustic mixer more nearly approx. the mirror image of the source sound. The adaptive filter has a modified LMS (least means square) algorithm governing the operation of its transversal filter which accommodates the inherent delays associated with the propagation of acoustic waves in a duct. A number of wave guides may connect the speaker to the acoustic mixer. A single electronic circuit may provide the signal processing for a pair of ducts.

US 4473906A

The active attenuator includes an input sensor for detecting the source vibration and a cancelling speaker for generating cancelling vibration. An error sensor is provided for sensing the combination of source and cancelling vibration and an electronic controller is coupled to the input sensor, cancelling speaker and error sensor.

The electronic controller includes an adaptive cancelling filter which employs a deterministic algorithm operable to accommodate the propagation delays of the vibration sensed by the input and error sensors. The controller produces an

output to activate and control the cancelling speaker for the production of cancelling vibration.

USE - Partic. for cancelling noise carried by heating and ventilating ducts and originating from machinery and mfr. operations. (27pp)

CHOSEN-DRAWING: Dwg.5/16 Dwg.5 Dwg.5

TITLE-TERMS: ACOUSTIC ATTENUATE ACTIVE SOUND CANCEL ADAPT FILTER DRIVE SPEAKER PRODUCE CANCEL SOUND DEGREE PHASE

DERWENT-CLASS: P86 Q51 Q67 W04 X27

EPI-CODES: W04-G; W04-T; X27-E01;